Dialogic®

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User's Guide

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Revision History

This revision history summarizes the changes made in each published version of this document.

Document No.	Publication Date	Description of Revisions
64-0260-02	September 2007	Updated to support Version 5.1 SU2 Software. Added new DMG1004LS model. Added note to say H.323 is only supported in Version 5.1 SU1 Software or earlier. Chapter 3, "Parameter Reference" added new parameter in Serial Protocol Parameters section for MCI Message Type. Chapter 3, "Parameter Reference" added new parameters in Gateway Advanced Group section for RTP Source IP Address Validation, RTP Source UDP Port Validation, SNMP Community Name, SNMP System Name, SNMP System Contact, SNMP System Location, and Telnet Server Enabled. Chapter 3, "Parameter Reference" added new parameter in Session Initiation Protocol (SIP) Parameters section for DNS Translation of Phone Numbers. Chapter 5, "Dial Plan" added ability to manipulate Redirecting Party in the CPID (Number) Manipulation section.
64-0260-01	April 2007	Updated to support Version 5.1 SU1 Software. Global Updates: Changed product names and links from Intel to Dialogic. The document number and title have been updated with new naming conventions. The products previously known as Intel NetStructure PBX-IP Media Gateway (PIMG) and T1/E1-IP Media Gateway (TIMG) are now Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000). For more product name changes, refer to New Product Naming Conventions. Updated notes to say LAN2 is only supported in Version 5.1 SU1 Software or later. Chapter 3, "Parameter Reference" added new parameters Client IP Address, Client Subnet Mask, IP Management Interface, Ethernet Interface, Destination Address, Destination Mask, Gateway Address, Call Progress Filter SNR in dB (DMG2000 Only), and Call Progress Filter Two Tones Max Twist in dB (DMG2000 Only). Chapter 5, "Dial Plan" added new chapter. Chapter 8, "Unit Status" added Section 8.5.1, "SNMP Agent", on page 184, Section 8.5.2, "Supported MIBs", on page 185, and Section 8.5.3, "SNMP Alarms", on page 187. Chapter 9, "Diagnostics" added Section 9.1, "VoIP Interface Test", on page 195, Section 9.2, "PBX/PSTN Interface Test", on page 199, and Section 9.3, "PBX Self Verification Test", on page 203.
C73213-002 Rev 01	November 2006	Updated to support Version 5.1 Software. Chapter 1, "Overview" added information about the DMG2120DTI model and added "Security" section. Chapter 3, "Parameter Reference" added various new parameters and added "Security Parameters" section. Chapter 7, "Data Security" added new chapter. Removed the PBX configuration chapters from the User's Guide.
C73213-001 Rev 01	April 2006	Initial version of document to support Version 5.0 Software.

About This Publication

The following topics provide information about this guide:

- Purpose
- Intended Audience
- How to Use This Publication
- Related Information

Purpose

This document provides information about installing, configuring, operating, and maintaining the Dialogic® Media Gateway.

Intended Audience

This information is intended for:

- Distributors
- System Integrators
- Value Added Resellers (VARs)
- Original Equipment Manufacturers (OEMs)

How to Use This Publication

This information is organized as follows:

- Chapter 1, "Overview" provides a description of the product and discusses call routing, address translation, and the Web interface.
- Chapter 2, "Configuring the Media Gateway" provides procedures for configuring and upgrading the Media Gateway.
- Chapter 3, "Parameter Reference" lists the Media Gateway parameters that can be configured from the Web interface.
- Chapter 4, "Working with Call Progress Tones" describes how to view, edit, learn, and validate call progress tones from the Web interface.
- Chapter 5, "Dial Plan" describes a set of rules used to define the characteristics of a call routed through the Media Gateway.
- Chapter 6, "Media Gateway Parsers" describes the Media Gateway in-band Type I (on-hook) and Type II (off-hook) integration parsers for analog units, and the display parser for digital

units. These parsers allow the user to define the meaning of either the in-band/on-hook integration strings or display strings received from the telephony network.

- Chapter 7, "Data Security" provides information about configuring security on the Media Gateway for HTTP, call control, and voice.
- Chapter 8, "Unit Status" describes the various types of status information that may be obtained about the Media Gateway.
- Chapter 9, "Diagnostics" provides information about using diagnostic logging, running the PBX self test, and using the various terminal commands to perform diagnostics on the Media Gateway.

Note: The products previously known as Intel NetStructure PBX-Media Gateway and T1/E1-Media Gateway are now Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000). For more product name changes, refer to New Product Naming Conventions.

New Product Naming Conventions

Previous Name	New Name
PBX-IP Media Gateway	Dialogic® 1000 Media Gateway (DMG1000)
PIMG	DMG1000
PIMG40LS	DMG1004LS
PIMG80LS	DMG1008LS
PIMG80DNI	DMG1008DNI
PIMG80MTLDNI	DMG1008MTLDNI
PIMG80RLMDNI	DMG1008RLMDNI
T1/E1-IP Media Gateway	Dialogic® 2000 Media Gateway (DMG2000)
TIMG	DMG2000
TIMG300DTI	DMG2030DTI
TIMG600DTI	DMG2060DTI
TIMG1200DTI	DMG2120DTI

Related Information

For additional information related to the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) products, see the following:

- *Dialogic® 1000 and 2000 Media Gateway Series Getting Started Guide* for information about installing, cabling, and initializing the product prior to performing configuration and operation tasks.
- *Dialogic® Media Gateway Installation and Configuration Integration Notes* for details on typical installation and configuration of Media Gateway when used to interface between PBX and unified messaging application. Available at: *http://www.dialogic.com/support/helpweb/mg/integration.htm*
- *http://www.dialogic.com/support/* (for technical support)

- *http://www.dialogic.com/* (for product information)

Overview 1

The following information provides an overview of the Dialogic® Media Gateway:

- Product Description 21
- Call Routing in Phone Emulating Mode 26
- Voice over IP Address Translation 27
- Security 28
- Web Interface. 29
- Online Help 33

1.1 Product Description

The Media Gateway is a telephony gateway appliance that connects to phone lines through its telephony interface and connects to a LAN via a 10 BaseT or 100 BaseT Ethernet connector.

NOTE: Throughout this document, the term **Media Gateway** addresses information that applies to both the DMG1000 and DMG2000 products. The term **DMG1000** applies only to information relating to the DMG1000 product and the term **DMG2000** applies only to information relating to the DMG2000 product.

The Media Gateway provides an inexpensive bridge between a legacy PBX or public switched telephone network (PSTN) and a managed packet network. This device converts signals from circuit switched equipment into either H.323 or Session Initiation Protocol (SIP) standard protocol for transmission over a local area network (LAN) or wide area network (WAN) to communications devices such as IP phones, wireless phones, and IP servers in almost any location.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

The DMG1000 is available in the following models:

- **DMG1008MTLDNI** - Supports phone emulation mode for Mitel digital PBXs.
- **DMG1008DNI** - Supports phone emulation mode for a number of digital PBXs, including Avaya, Nortel, NEC, and Siemens.
- **DMG1008RLMDNI** - Supports phone emulation mode for Rolm 8000 and 9751 switches.
- **DMG1008LS, DMG1004LS** - Supports phone emulation mode for analog interfaces.

The DMG2000 is available in the following models:

- **DMG2030DTI** - Supports phone emulation mode for a single T1 or E1 interface.
- **DMG2060DTI** - Supports phone emulation mode for two T1 or E1 interfaces.
- **DMG2120DTI** - Supports phone emulation mode for four T1 or E1 interfaces.

Note: The Item Market Name on the Media Gateway may vary slightly, depending on the version.

Depending on the model, the Media Gateway can be configured for the following operating mode:

- Phone Emulating

1.1.1 Phone Emulating

In the Phone Emulating mode, the Media Gateway operates as a telephony gateway appliance that emulates:

- up to eight station sets (DMG1000 models)
- up to 24 station sets (single T1 DMG2000 model)
- up to 48 station sets (dual T1 DMG2000 model)
- up to 96 station sets (quad T1 DMG2000 model)
- up to 30 station sets (single E1 DMG2000 model)
- up to 60 station sets (dual E1 DMG2000 model)
- up to 120 station sets (quad E1 DMG2000 model)

for transporting PBX functionality over a packet-switched network.

The Media Gateway translates protocols for call setup and release between the IP network and the PBX or PSTN, and converts the media formats between the two networks.

Figure 1 shows how the Media Gateway provides a gateway between voice over IP (VoIP) devices (H.323 or SIP) on a LAN and the PBX. By emulating station sets to the proprietary PBX, the Media Gateway provides full call party information to the IP network.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

The DMG2000 or the analog version (DMG1008LS, DMG1004LS) of the DMG1000 can also connect directly to the PSTN, as shown in Figure 2.

Figure 1. Typical IP Gateway Phone Emulating Topology - PBX Connection

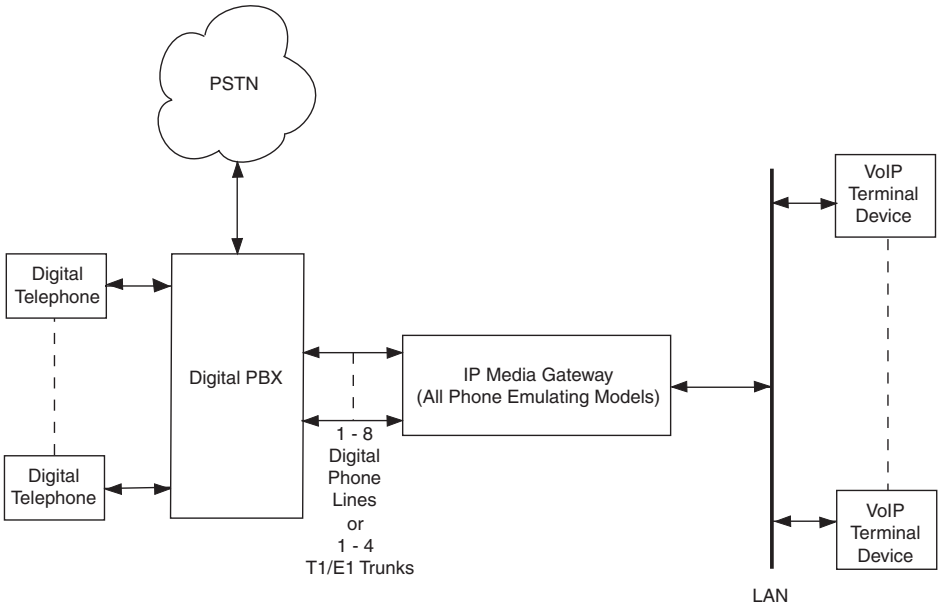
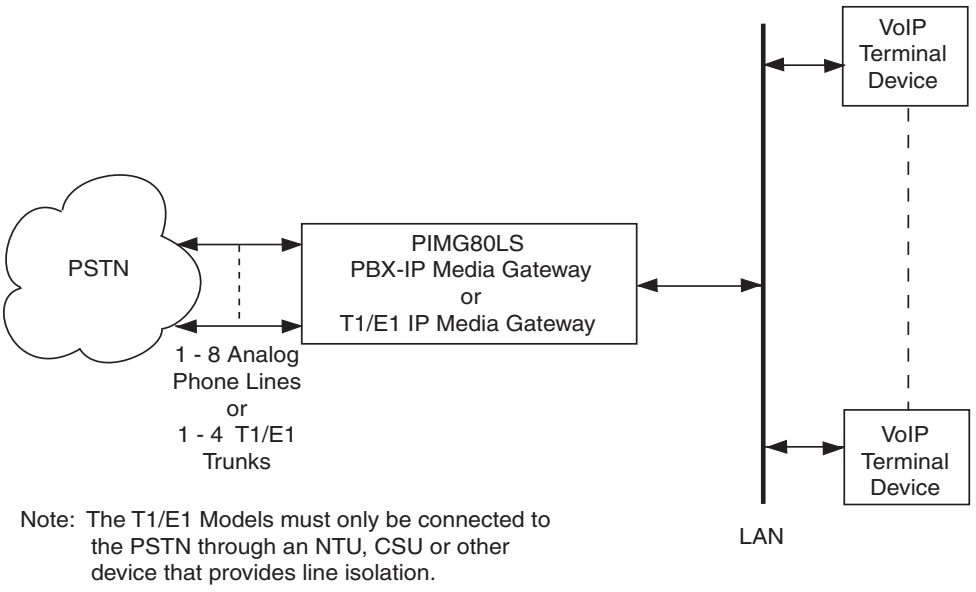


Figure 2. Typical IP Gateway Phone Emulating Topology - PSTN Connection


PSTN
1 - 8 Analog Phone Lines or 1 - 4 T1/E1 Trunks
PIMG80LS PBX-IP Media Gateway or T1/E1 IP Media Gateway
VoIP Terminal Device
VoIP Terminal Device
LAN

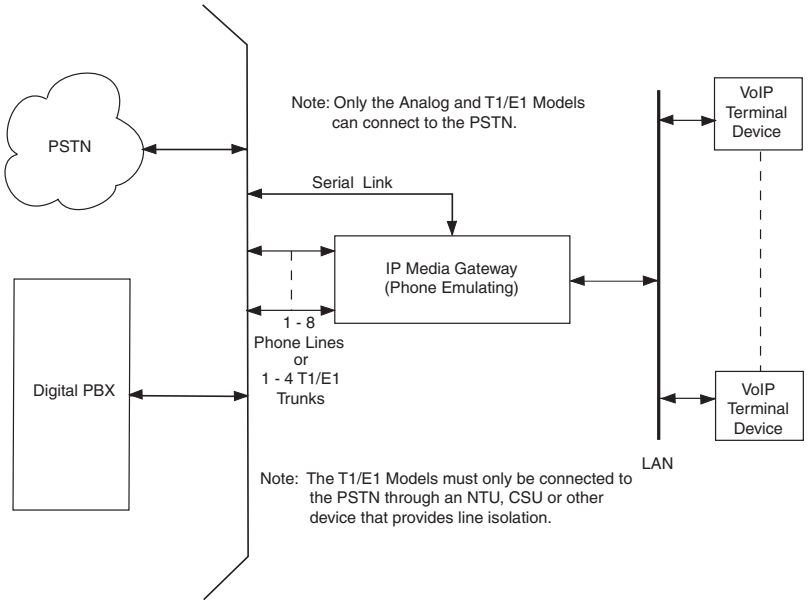
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

1.1.1.1 Serial Protocol Support in Phone Emulating Mode

By emulating telephone sets to the switch, the Media Gateway provides call party information over the IP network. However, the amount of call party information that the Media Gateway can provide is limited to the amount of data that the switch provides its station sets. Some proprietary switches provide full call party information across the station set interfaces while others provide little or no call party information across the station set interface. The PBX switches that provide little or no call party information typically will provide full call party information across a separate serial interface connection.

For this reason, the Media Gateway supports a serial link interface to the switch or PSTN, and supports several serial protocols. Figure 3 shows how the Media Gateway connects to a switch or PSTN that uses a serial link to provide call party information. Using this serial link, the Media Gateway is able to provide full call party information on a PBX switch or PSTN that provides little or no call party information via their station set interfaces.

Figure 3. IP Gateway Using Serial Link


PSTN
Note: Only the Analog and T1/E1 Models can connect to the PSTN.
VoIP Terminal Device
Serial Link
IP Media Gateway (Phone Emulating)
1 - 8 Phone Lines or 1 - 4 T1/E1 Trunks
Digital PBX
VoIP Terminal Device
LAN
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

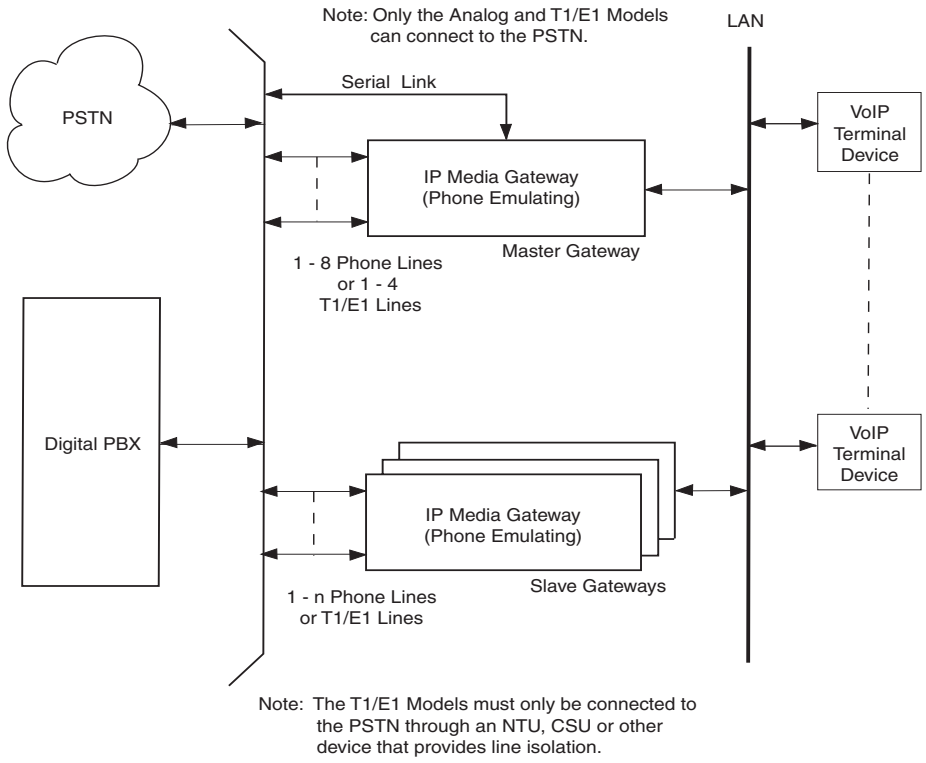
When a call arrives at a telephony port on the Media Gateway, the switch or PSTN will send a data packet across the serial link containing the call party information associated with the call. If configured to use the serial interface, the Media Gateway will use the data in the serial packet as the call party information when the call notification is sent across the IP network to the VoIP terminal device that the call is intended for - superseding any call information that may have arrived across the station set interface. Finally, the serial protocols also support the message waiting indication (MWI) feature that allows the Media Gateway to control message indications on telephone sets connected to the switch or PSTN.

For DMG1000 types, the serial link from the switch is connected to the serial port interface on the DMG1000 (DIAGNOSTICS connector). Table 1, "DIAGNOSTICS Connector Pin Designations" in the *Getting Started Guide* shows the connector pin designations.

For DMG2000 types, the serial link from the switch is connected to the serial port interface on the DMG2000 (COM 1 or COM 2 connector). Table 4, "COM 1 and COM 2 Connector Pin Designations" in the *Getting Started Guide* shows the connector pin designations.

The switch only provides a single serial link connection. At a site where there are multiple Media Gateway units, only one of the units can be physically connected to the switch or PSTN serial link. This unit is the serial protocol Master Media Gateway while the remaining units are considered serial protocol slaves. Figure 4 shows how multiple Media Gateways connect to a switch or PSTN that uses a serial link to provide call party information. It is the responsibility of the Master gateway to send all serial link data intended for Slave Gateways to the Slave Gateways across the IP network. Similarly, anytime a Slave Gateways needs to communicate to the switch or PSTN across the serial link, the slave unit sends the data across the IP link to the Master Gateway. The Master Gateway will then send the data across the serial link on behalf of the slave device.

Figure 4. Multiple IP Gateways Using Serial Link


Note: Only the Analog and T1/E1 Models can connect to the PSTN.
LAN
PSTN
Serial Link
VoIP Terminal Device
IP Media Gateway (Phone Emulating)
Master Gateway
1 - 8 Phone Lines or 1 - 4 T1/E1 Lines
Digital PBX
VoIP Terminal Device
IP Media Gateway (Phone Emulating)
Slave Gateways
1 - n Phone Lines or T1/E1 Lines
Note: The T1/E1 Models must only be connected to the PSTN through an NTU, CSU or other device that provides line isolation.

The Media Gateway supports the following serial protocols:

- SMDI (Simple Message Desk Interface)
- MCI (NEC Systems only)
- MD110 (Ericsson Systems only)

Both the serial interface and the selection of which serial protocol to use are configurable using the Web interface. Refer to Chapter 3, "Parameter Reference" for information about configuring the serial interface and serial protocols.

1.2 Call Routing in Phone Emulating Mode

The Media Gateway routes calls from the Switch network to a VoIP destination on the IP network. Conversely, it routes calls from the IP network through a Switch port to a destination telephone number on the Switch network. The Media Gateway supports the following call routing options:

- User configurable list of VoIP Servers
- IP load Balancing
- IP Fault Tolerance

In its simplest form, call routing is supported by configuring a single VoIP Server to receive and/or originate calls through the Media Gateway. In this manner, all inbound Switch-to-IP calls will be sent to the single user configured VoIP endpoint. For IP-to-Switch calls, the telephony port will be selected in a round robin fashion – where each IP-to-Switch call will be routed to the next available telephony port.

If the user configures more than one VoIP Server to receive and/or originate calls through the Media Gateway, then the user has the option to have the incoming Switch-to-IP calls load balanced between the configured VoIP Servers. Specifically, incoming Switch calls will be routed to a VoIP server in a round-robin fashion. For example, if there are three (3) VoIP Servers configured, the first call will be routed to the first VoIP server, the second call will be routed to the second VoIP server, the third call will be routed to the third VoIP server. The next call will be routed to the first VoIP Server and the process will start all over again. IP-to-Switch calls are handled in the same way when multiple VoIP Servers are configured as when only a single VoIP server is configured (i.e. in a round robin fashion).

If the user configures more than one VoIP Server to receive and/or originate calls through the Media Gateway, then the user has the option to support fault tolerance on the incoming Switch-to-IP calls. Specifically, if the VoIP server fails to respond to incoming Switch call (or responds with an error), the Media Gateway will route the call to the next VoIP Server in the user configurable list of VoIP Servers.

Note: If both IP Load Balancing and Fault Tolerance are enabled, then incoming IP-to-Switch calls will be routed to the configured VoIP Servers in a round robin fashion and, if at any time a VoIP Server fails to respond or responds with any error, the Media Gateway will route the call to the next available VoIP Server.

1.2.1 Un-Routable Calls

A call is un-routable by the Media Gateway if the unit is unable to route the call to the other network. This may occur if there is insufficient destination address information to determine a destination for the call, or if there are not enough free resources on the Media Gateway to route the call. The Media Gateway provides a number of different ways to handle these conditions in order to insure that no calls are dropped or not completed.

1.2.1.1 IP to Switch Calls

There are a few circumstances where an inbound IP call may not be routed to the Switch network. They are the following:

- No available Switch Ports
  - There are no available Media Gateway Switch ports on which to carry the call.
- Invalid or No Switch destination address specified
  - The IP call information does not contain a valid E.164 destination address for the Switch network.

The administrator may specify a default destination IP address that is to receive any inbound IP calls that cannot be routed to the Switch. If the administrator specifies a default destination IP address for un-routable IP calls, then any inbound IP call that cannot be routed to the Switch network is forwarded to the specified default IP address. If the administrator does not specify a default destination IP address, then any inbound IP call that cannot be routed to the Switch is rejected by the Media Gateway.

1.2.1.2 Switch to IP Calls

There are a few circumstances in which an inbound Switch call may not be routed to the IP network. They are the following:

- IP destination not configured.
- IP destination not present.

The administrator may specify a default destination Switch extension that is to receive any inbound Switch calls that cannot be routed to the IP network. If the administrator specifies a default destination Switch extension for un-routable Switch calls, then any inbound Switch call that cannot be routed to the IP network is transferred to the specified default Switch extension. If the administrator does not specify a default destination Switch extension, then any inbound Switch call that cannot be routed to the IP network is ignored by the Media Gateway.

1.3 Voice over IP Address Translation

The Voice over IP (VoIP) Address Translator provides network services to H.323 or SIP devices such as the Media Gateway. H.323 devices register with the VoIP Address Translator to send and

receive H.323 calls. SIP devices register with the VoIP Address Translator to sent and receive SIP calls.

Note: In H.323 mode, the VoIP Address Translator is a Gatekeeper. In SIP mode, the VoIP Address Translator is a SIP Registration Server and/or a SIP Proxy Server.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

The VoIP Address Translator can provide network services such as:

- Controlling the number and type of connections allowed across the network.
- Helping to route a call to the correct destination.
- Determining and maintaining the network address for incoming calls.

Without a VoIP Address Translator, all IP destination addresses must be specified to the Media Gateway as IP v4 addresses (ex. 10.10.4.128). IP terminal devices must also explicitly specify the IP address of the Media Gateway as the desired gateway when originating PBX calls.

With a VoIP Address Translator, IP destination addresses may be specified to the Media Gateway as e-mail addresses, alphanumeric aliases, E.164 telephone numbers, domain names, and any other format supported by the VoIP Address Translator. IP terminal devices may explicitly specify the IP address of the Media Gateway as the desired gateway, or they may rely on the VoIP Address Translator to address the Media Gateway using the gateway prefix configured on the Media Gateway.

1.4 Security

The Media Gateway supports two kinds of security:

- Access Security
- Data Security

1.4.1 Access Security

Access to the Media Gateway is secured by requiring a user-name and password to login to the gateway. The user-name and password are required for all interfaces of the Media Gateway (Web interface, serial interface, and telnet interface).

1.4.2 Data Security

Data security for the Media Gateway includes the use of various secure protocols when transmitting and receiving data. The Media Gateway supports security for three types of data:

- HTTP security - Data transmitted between the Media Gateway and a Web browser. To secure HTTP, the Media Gateway uses HTTPS protocol.
- Call Control security - Data used to setup and tear down a call. To secure Call Control, the Media Gateway uses Transport Layer Security (TLS) on top of SIP.

- Voice security - The actual conversation once a call is connected. To secure voice, the Media Gateway uses SRTP.

Note: Security is not provided over H.323. If H.323 is selected instead of SIP, TLS and SRTP are not supported and call control and voice data cannot be secured.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

Because, for security, the HTTPS and TLS protocols also require digital identity certificates (e.g. public key certificates), Certificate Configuration and Management is also provided by the Media Gateway.

For additional information about how the Media Gateway supports security, see Chapter 7, "Data Security".

1.5 Web Interface

The Web interface is accessed from a workstation on the Ethernet connected to the **LAN** connector on the rear panel of the Media Gateway. Communication is established by starting a Web browser at the workstation and entering the IP address of the Media Gateway. The unit's Web interface is password protected and the password can be changed by the system administrator. Figure 5 shows a typical Web interface page for a DMG1000 unit and Figure 6 shows a typical Web interface page for a DMG2000 unit.

Figure 5. DMG1000 Web Interface

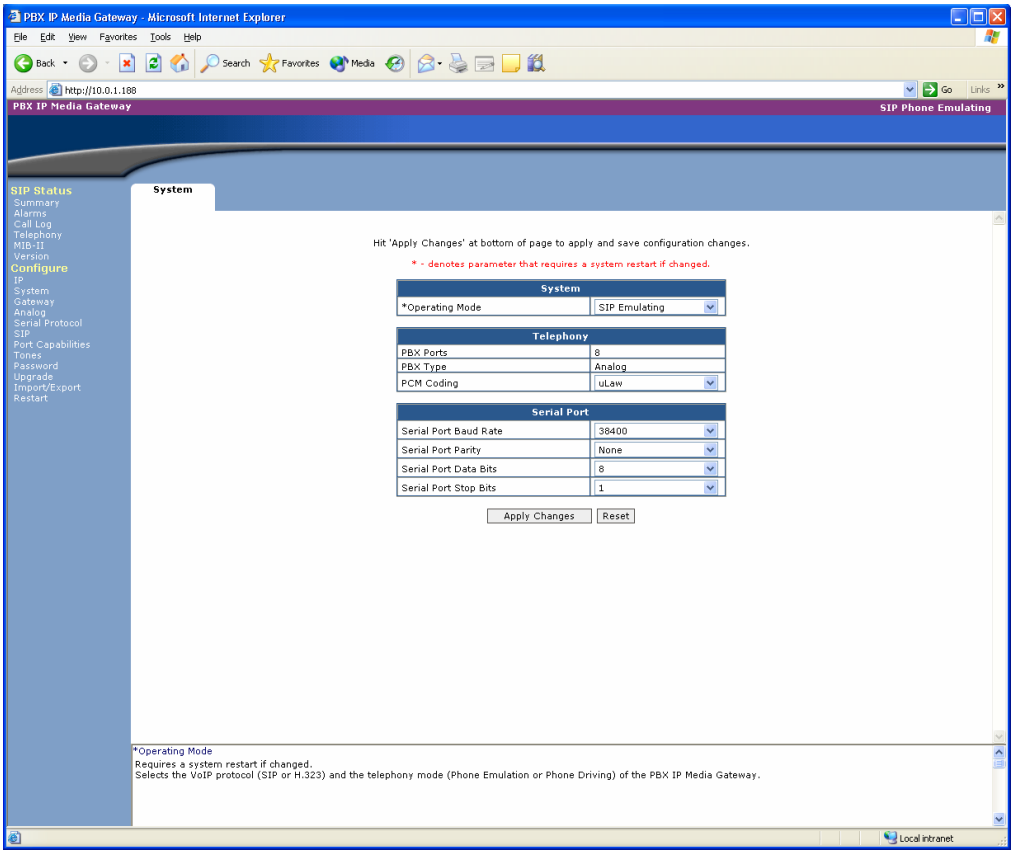
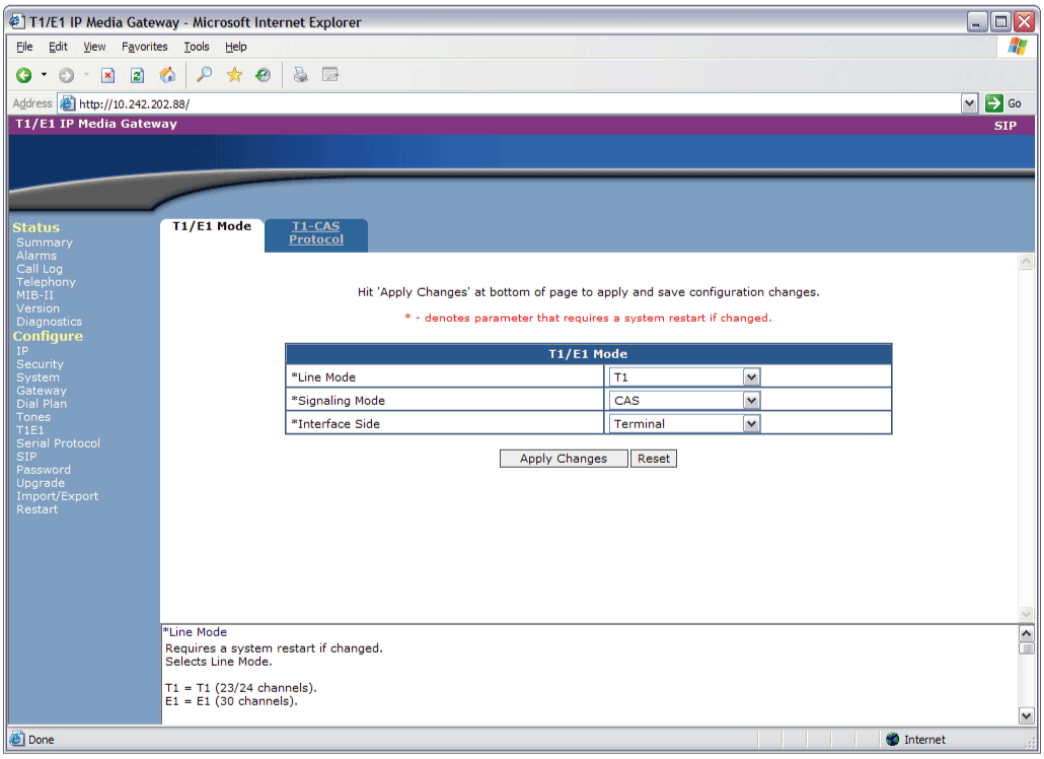


Figure 6. DMG2000 Web Interface



The Media Gateway's Web interface is divided into two sections, **H.323** or **SIP Status** and **Configure**. The Status Web pages provide run-time information and statistics about the operation of the unit.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

The Configure Web pages provide the system administrator with an interface to configure the Media Gateway. The Configuration menu selections include:

- **IP** - Selecting IP brings up the IP Web page which allows you to configure the IP parameters. For detailed information about the IP parameters, see Section 3.1, "IP Settings", on page 41.
- **Security** - Selecting Security brings up the Security Web page which allows you to configure Security parameters. For detailed information about the Security parameters, see Section 3.2, "Security Parameters", on page 47.
- **System** - Selecting System brings up the System Web page which allows you to configure System level parameters. For detailed information about the System parameters, see Section 3.3, "System Parameters", on page 51.
- **Analog (Models DMG1008LS, DMG1004LS only)** - Selecting Analog displays the Analog Web page which allows you to configure the Analog parameters. These parameters only apply to the analog version of the DMG1000. See Section 3.4, "Analog Parameters (Models DMG1008LS, DMG1004LS Only)", on page 55 for detailed information about the Analog parameters.

- **Serial Protocol (Phone Emulating mode only)** - Selecting Serial Protocol brings up the Serial Protocol Web page. This Web page allows you to configure the serial protocol for the Media Gateway serial port. See Section 3.5, "Serial Protocol Parameters", on page 62 for detailed information about the Serial Protocol parameters.
- **Gateway** - Selecting Gateway brings up the Gateway Web page which allows you to configure Gateway parameters. Some of these parameters differ, depending on whether the unit is being operated in the Emulating mode or the Driver mode. For detailed information about the Gateway parameters, see Section 3.6, "Gateway Parameters", on page 66.
- **T1E1 (DMG2000 models only)** - Selecting T1E1 allows you to configure the T1 and E1 parameters for the T1 or E1 ports on a DMG2000 unit. For detailed information about the T1 and E1 parameters, see Section 3.7, "T1/E1 Parameters (DMG2000 Only)", on page 87.
- **H.323** - If you have chosen either H.323 Emulating or H.323 Driving as the operating mode, selecting H.323 brings up the H.323 Web page which allows you to configure H.323 parameters. For detailed information about the H.323 parameters, see Section 3.8, "H.323 Parameters", on page 99.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

- **SIP** - If you have chosen either SIP Emulating or SIP Driving as the operating mode, selecting SIP brings up the SIP Web page which allows you to configure SIP parameters. For detailed information about the SIP parameters, see Section 3.9, "Session Initiation Protocol (SIP) Parameters", on page 103.
- **Tones** - Selecting Tones allows you to manipulate call progress tone parameters. These parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Media Gateway detects during call progress analysis. For more information, see Section 3.11, "Tone Parameters", on page 125 and Chapter 4, "Working with Call Progress Tones".
- **Password** - Selecting Password brings up the Password Web page. This page allows you to change your password. For information about the procedure for changing your password, see Section 2.3, "Changing the Password", on page 37.
- **Upgrade** - Selecting Upgrade brings up the Upgrade Web page. The Upgrade Web page allows you to upgrade the Media Gateway software. For information about the procedure for upgrading the software, see Section 2.7, "Upgrading the Software", on page 40.
- **Import/Export** - Selecting Import/Export brings up the Import/Export Web page which allows you to save a copy of your current configuration (Export) to a directory on the computer or to import a configuration file to the Media Gateway (Import). For information about exporting a configuration file, see Section 2.6.1, "Exporting Configuration Information", on page 39. For information about importing a configuration file, see Section 2.6.2, "Importing Configuration Information", on page 39.
- **Restart** - Selecting Restart from the Configure menu brings up the Restart Web page which allows you to restart the Media Gateway. Restarting the unit is required when certain parameter values are changed. You have two options to choose from when restarting the unit:
  - Restart Unit Now - Clicking this button will cause the unit to restart immediately.
  - Restart Unit When Idle - Clicking this button will cause the unit to restart when the unit is considered in the idle state. In the idle state, there are no calls (incoming, outgoing, or connected) on any of the PBX ports of the Media Gateway. By selecting this option, you will schedule a restart time that minimizes the effect the restart will have on call traffic through the unit.

1.6 Online Help

Context-sensitive online Help is provided for the various Web pages. When you move the mouse pointer over a parameter name, the pointer changes from an arrow to an insertion symbol and Help information about that parameter is displayed at the bottom of the page.

Configuring the Media Gateway 2

Information about configuring, and upgrading the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) is contained in the following sections:

- Setting the IP Address 35
- Basic Configuration Via the Serial Port 36
- Changing the Password 37
- Configuration Procedure 37
- Restart Options 38
- Importing and Exporting Configuration Information 38
- Upgrading the Software 40

Note: Refer to the *Getting Started Guide* for information about initially logging on to the Media Gateway and performing basic configuration via the serial port.

2.1 Setting the IP Address

Once the IP address of the Media Gateway unit has been configured through the Web interface, the unit must be restarted to activate the new configuration. The Media Gateway unit will then be accessible to the Web browser at the newly configured IP address. All configuration parameters are saved in the Media Gateway's non-volatile memory.

Use the following steps to assign the Media Gateway a unique IP address:

1. Select the IP configuration Web page from the **Configure** menu on the left side of the Web page.
2. Change the unit's IP address from the default address by entering the new IP address in the **Client IP Address** box.
3. Configure the subnet mask if it is different from the default value by entering the new subnet mask in the **Client Subnet Mask** box.
4. Configure the IP address of the default network gateway router by entering the IP address in the **Default Network Gateway Address** box.
5. Click on the **Apply Changes** button to save the configuration in the database.

Note: Refer to Chapter 3, "Parameter Reference" for a description of all of the configuration parameters that may be changed.

6. For the configuration change to take effect, you will be prompted to restart the Media Gateway by clicking on Restart on the Web page or by selecting **Restart** from the **Configuration** menu.
7. When the **Restart** Web page appears, click on **Restart Unit Now** to restart the Media Gateway.

8. Once the system completes its initialization (after approximately one minute), browse to the new IP address. The Web browser can now access the unit at the new IP address.
9. If it was necessary to change the IP address of your Windows® workstation in the Initial Log On procedure (described in the *Getting Started Guide*), you should now change it back to the original IP address and access the Media Gateway using the Media Gateway's newly configured IP address.
10. You must now log on to the system again. After logging on, the Summary Web page will now appear. You may now select any item from the **Status** or **Configure** menu.

2.2 Basic Configuration Via the Serial Port

When the default IP address is unreachable (or if a previously configured IP address becomes unreachable) on your network, use the following procedure to set the Client IP Address, Client Subnet Mask, Default Network Gateway Address, Operating Mode, and PBX Type parameters of the Media Gateway:

1. Connect a serial cable to the serial connector on the rear panel of the Media Gateway unit (DIAGNOSTICS connector on the DMG1000 models or the COM 2 connector on the DMG2000 models). For connector pin designation information, refer to the DIAGNOSTICS Connector Pin Designations table or the COM 1 and COM 2 Connector Pin Designations table in the *Getting Started Guide*.
2. Using a standard serial interface application (for example, Procomm Plus or HyperTerminal), set the workstation to the following:
   - Baud Rate = 38400 (DMG1000) or 115200 (DMG2000) bps
   - Parity = None
   - Data Bits = 8
   - Stop Bits = 1
   - Hardware Flow Control = Off
3. Press the Enter key repeatedly until the following prompt appears:

   PIMG>
4. At the prompt, type **pwd** and press Enter.
5. When prompted, enter the password for the admin user (the default is **IpodAdmin**) and press Enter.
6. At the prompt, type **quickcfg** and press Enter.
7. You will then be prompted to enter the following Media Gateway parameter information:
   - Client IP Address (See Client IP Address parameter information in Section 3.1, "IP Settings", on page 41.)
   - Client Subnet Mask (See Client Subnet Mask parameter information in Section 3.1, "IP Settings", on page 41.)
   - Network Gateway IP (if required) (See Default Network Gateway Address parameter information in Section 3.1, "IP Settings", on page 41.)
   - Select Operating Mode (See Operating Mode parameter information in Section 3.3.2, "Telephony Group", on page 51.)

- For DMG1000 models, select PBX Type (See PBX Type parameter information in Section 3.3, "System Parameters", on page 51.)
- For DMG2000 models, select:
  - Line Mode (Section 3.7.1.1, "Line Mode", on page 87)
  - CAS Protocol (Section 3.7.1.2, "Signaling Mode", on page 87 and Section 3.7.2.1, "T1 CAS Protocol", on page 88) or ISDN Protocol (Section 3.7.1.2, "Signaling Mode", on page 87 and Section 3.7.3.1, "ISDN Protocol", on page 94)

8. When prompted that the parameters have been successfully configured, type **restart** at the PIMG> prompt to restart the Media Gateway.

   You should now be able to connect to the Media Gateway from the Web Interface using the newly configured IP Address.

2.3 Changing the Password

The steps for changing the password are described in the following procedure:

1. Start your Web browser.
2. In the Web browser address box, enter the IP address of the Media Gateway that you wish to access.
3. When the **System Login** Web page appears, enter the user name and current password in the boxes provided and click on the **Log On** button.

Note: The user name and password are case sensitive.

4. Once the login has been accepted, the Media Gateway Status/Configure Web page will appear. Select the **Password** Web page from the **Configure** menu on the left side of the page.
5. Enter the current password in the Old Password box.
6. Enter the new password in the New Password box.
7. Enter the new password a second time in the Confirm box. Then, click on the **Change** button to replace the old password with the new password.

2.4 Configuration Procedure

The Media Gateway is configured using a Web interface.

To change configuration parameter values through the Web interface, use the following procedure:

1. From a workstation connected to the Media Gateway via the Ethernet, start the Web browser.
2. Enter the IP address assigned to the Media Gateway. For example:

   ```
   http://10.12.13.74
   ```

3. When the System Login Web page appears, enter **admin** in the User Name box and your password in the Password box, then click on the **Log On** button.

Note: The user name and password are case sensitive.

4. Select the appropriate configuration Web page from the **Configure** menu. For example, to define the PCM Coding as aLaw, select the **System** Web page from the menu.

   When the **System** Web page appears, select **aLaw** from the drop down box as the PCM Coding parameter value.

5. Click on the **Apply Changes** button to save the new configuration in the database, or click on the **Reset** button to return the parameter to the previous value.

   Note: If the parameter whose value you are changing requires a restart for the change to take effect, you will be prompted by a flashing **Restart Required** blue box in the upper left-hand corner of the screen.

6. If a restart is required, click on Restart on the System screen or select **Restart** from the **Configuration** menu.

7. When the **Restart** Web page appears, click on **Restart Unit Now** or **Restart Unit When Idle** to restart the Media Gateway. (See Section 2.5, "Restart Options", on page 38).

8. Once the system completes its initialization (after approximately one minute), select **Refresh** from the Web Browser **View** Menu.

9. You must now log on to the system again. After logging on, the Summary screen will appear. You may now select any item from the **Status** or **Configure** menu.

2.5 Restart Options

Selecting Restart from a configuration screen or selecting **Restart** from the **Configure** menu brings up the **Restart** Web page which allows you to restart the Media Gateway. Restarting the unit is required when certain parameter values are changed. You have two options to choose from when restating the unit:

- **Restart Unit Now** - Clicking this button will cause the unit to restart immediately.
- **Restart Unit When Idle** - Clicking this button will delay the unit from restarting until the unit is considered in the idle state. In the idle state, there are no calls (incoming, outgoing, or connected) on any of the PBX ports of the Media Gateway. By selecting this option, you will schedule a restart time that minimizes the effect the restart will have on call traffic through the unit.

2.6 Importing and Exporting Configuration Information

The Import/Export Web page allows you to import or export Media Gateway configuration information to a *.ini file on the host machine connected to the Media Gateway via the Web interface.

Scenarios where importing or exporting the configuration information may be useful include:

- **Backup configuration** - It is recommended that, after configuring the Media Gateway, you backup the configuration information by exporting it to a *.ini file. If at some time in the future the configuration has been changed and now you wish to return to the previous configuration, you can import the previous configuration data back to the Media Gateway.

- **Multiple unit implementation** - At a site where there are multiple Media Gateway units, much of the configuration will be the same between units (e.g. Audio Parameters, SIP/H.323 Parameters, etc.). An installer could configure the first unit, export the *.ini file, edit the unit specific parameters in the *.ini file (e.g. IP Address) using Notepad, and then import the modified file to the next unit. This will speed up and simplify the installation procedure at larger sites.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

- **Support** - If the Web interface is inaccessible to customer support, you can export the configuration information to a *.ini file and then email the file to customer support for them to view the configuration directly.
- **Software upgrades** - Normally, software upgrades will not affect the configuration. When a major software upgrade is performed, however, existing configuration information may be erased and default settings restored. In this scenario, the current configuration information could be exported before the upgrade is performed and then imported back to the Media Gateway after the upgrade. All of the previous configuration data will then be restored, except for the new parameters, which would be at their default settings.

2.6.1 Exporting Configuration Information

To export the current configuration information:

1. Select the **Import/Export** Web page from the **Configure** menu.
2. Click the **Export Settings** button in the Export box. The File Download dialog box then appears.

   You may click the **Open** button to view/edit the configuration file in Notepad or click the **Save** button to download and save the file to a directory. The default name for the file is *config.ini*.

   Clicking the **Save** button causes the Save As dialog box to appear.
3. To save the configuration file to a directory, click the **Save** button. In the Save As dialog box, choose the directory in which you wish to save the configuration file and, if you wish, rename the file. Do not, however, change the file type from INI.

2.6.2 Importing Configuration Information

To import a configuration file:

1. Select the Import/Export Web page from the **Configure** menu.
2. Enter an INI file name in the Import box or use the **Browse...** button to select an INI file.
3. Click the **Import Settings** button to import the configuration information to the Media Gateway.

2.7 Upgrading the Software

Software upgrades for the Media Gateway products will be made available on the Dialogic support Web site as needed. Contact technical support to obtain software upgrade files. Upgrade files are uploaded to the Media Gateway using the Web interface.

Caution: When a major software upgrade is performed, existing configuration information may be erased and the default values restored. It is recommended that before upgrading the software, you should export the current configuration information using the Export utility. Refer to Section 2.6, "Importing and Exporting Configuration Information", on page 38 for information about backing up the configuration information.

When an upgrade file is made available, follow these steps to upgrade the Media Gateway:

1. Start your Web browser.
2. Download the software upgrade file from the Dialogic Support Web site.
3. In the Web browser address box, enter the IP address of the Media Gateway that you wish to upgrade.
4. When the System Login Web page appears, enter the user name and current password in the boxes provided and click on the **Log On** button.

Note: The user name and password are case sensitive.

5. Once the login has been accepted, the Media Gateway Summary Web page will appear. Select the **Upgrade** Web page from the **Configure** menu on the left side of the page.
6. Enter the path and filename of the upgrade file or click on the **Browse...** button to select the upgrade file.
7. Click on the **Install** button to upload the file.

   The Media Gateway will respond when the upgrade has completed (approximately 10-20 seconds).
8. If there is more than one upgrade file, repeat steps 5 through 7 for each additional file.
9. Select **Restart** from the **Configure** menu on the left side of the page.
10. On the **Restart** Web page, click on the **Restart Unit Now** or **Restart Unit When Idle** button. (See Section 2.5, "Restart Options", on page 38).
11. The unit will now restart and re-initialize using the new software.

Parameter Reference 3

This section lists each Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) configuration parameter that may be changed using the Web browser **Configure** menu. Also listed are non-menu (hidden) parameters which are not accessible from the Configure menu. Changing non-menu parameters significantly modifies the operation of the Media Gateway. Included in this section is a description of each parameter, the allowed values, and, where applicable, the default value. The parameters are grouped into the following major categories:

- IP Settings 41
- Security Parameters 47
- System Parameters 51
- Analog Parameters (Models DMG1008LS, DMG1004LS Only) 55
- Serial Protocol Parameters 62
- Gateway Parameters 66
- T1/E1 Parameters (DMG2000 Only) 87
- H.323 Parameters 99
- Session Initiation Protocol (SIP) Parameters 103
- Non-Menu (Hidden) Parameters 112
- Tone Parameters 125
- Configuration File Parameters 129

3.1 IP Settings

The IP Settings include the following groups:

- IP Settings, LAN1
- IP Settings, LAN2 (DMG2000)

3.1.1 IP Settings, LAN1

The IP Settings, LAN1 group includes the following parameters:

- Client IP Address
- Client Subnet Mask
- Default Network Gateway Address
- BOOTP Enabled
- SNTP Server IP Address

3.1.1.1 Client IP Address

Description: Sets the IP address of the Media Gateway.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 10.12.13.74

INI File Parameter Name = ipClientAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.1.2 Client Subnet Mask

Description: Sets the subnet mask of the Media Gateway.

Allowed Values: Any valid IP mask.

Default Value = 255.255.255.0

INI File Parameter Name = ipSubnetMask

Note: Unit requires a restart if this parameter value is changed.

3.1.1.3 Default Network Gateway Address

Description: Sets the IP address of the default network gateway router.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 10.10.11.1

INI File Parameter Name = ipRouterAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.1.4 BOOTP Enabled

Description: When enabled, this parameter causes the Media Gateway to issue a BOOTP request on startup. A BOOTP/TFTP server may be configured to provide IP credentials, firmware upgrades, and a configuration INI file.

Note: Either a BOOTP/TFTP server or DHCP server may be used to respond to the Media Gateway BOOTP request.

Allowed Values:

- Yes = Parameter enabled - Media Gateway issues a BOOTP request on startup.
- No = Parameter disabled - Media Gateway does not issue a BOOTP request on startup.

Default Value = No

INI File Parameter Name = ipBootpEnabled

Note: Unit requires a restart if this parameter value is changed.

3.1.1.5 SNTP Server IP Address

Description: Simple Network Time Protocol (SNTP) server IP address. Used for SIP Transport Layer Security (TLS) data verification

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = No default value

INI File Parameter Name = ipSntpServerAddr

Note: Unit requires a restart if this parameter value is changed.

3.1.2 IP Settings, LAN2 (DMG2000)

The IP Settings, LAN2 group includes the following parameters:

- Client IP Address
- Client Subnet Mask

Note: Currently, LAN2 is only supported in Version 5.1 SU1 Software or later.

3.1.2.1 Client IP Address

Description: Sets the IP address of the Media Gateway second Ethernet port. This Ethernet port may be used as the Maintenance port depending on the "IP Management Interface" setting. A value of 0.0.0.0 disables the LAN2 Ethernet port.

Note: The LAN2 subnet must not overlap with the LAN1 subnet other wise there will be IP routing issues.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 0.0.0.0 (disabled)

INI File Parameter Name = ipClientAddrLan2

Note: Unit requires a restart if this parameter value is changed.

3.1.2.2 Client Subnet Mask

Description: Sets the subnet mask of the Media Gateway maintenance Ethernet port.

Allowed Values: Any valid IP mask.

Default Value = 255.255.255.0

INI File Parameter Name = ipSubnetMaskLan2

Note: Unit requires a restart if this parameter value is changed.

3.1.3 IP Advanced Parameters (DMG2000)

The IP advanced parameters allow an advanced network topology to be configured using the LAN interfaces. The administrator can configure which LAN interfaces provide access to the IP management protocols. The administrator can also configure custom routing rules that replace or supplement the functionality of the default network gateway address.

Management Interface

The management IP protocols are accessible on all LAN interfaces by default. These management IP protocols include HTTP, HTTPS, SNMP, Syslog and Telnet. An administrator may choose to limit access to the management protocols based on the LAN interface. This means that only certain networks and subnets can access them. The management interface can be bound to LAN1 only, LAN2 only or all LANs (both LAN1 and LAN2). For example, the management protocols may be configured to be accessible only on LAN2. This means that LAN1 would not be a management interface and the management protocols would not be accessible via LAN1. The protocols would only be accessible by accessing the gateway at the LAN2 IP address.

The management interface group includes the parameter:

- IP Management Interface

3.1.3.1 IP Management Interface

Description: Sets which LAN interface is accessible for management protocols.

Allowed Values:

- LAN1 = Management functionality available on LAN1 interface only
- LAN2 = Management functionality available on LAN2 interface only
- ALL = Management functionality available on both LAN1 and LAN2 interfaces

Default Value = ALL

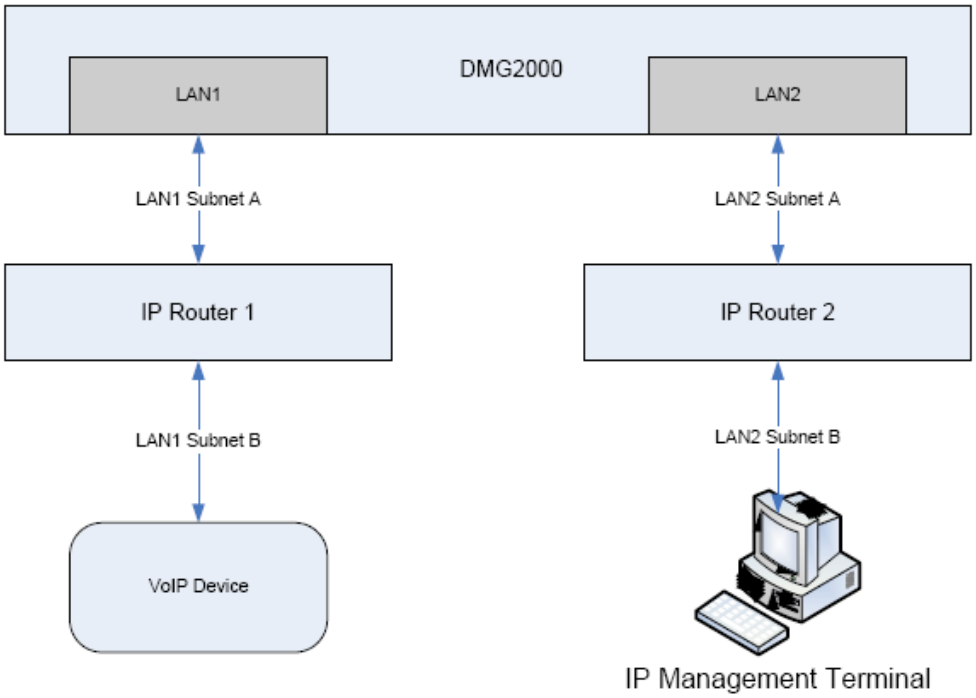
INI File Parameter Name = ipMgmt

Note: Unit requires a restart if this parameter value is changed.

3.1.3.2 Advanced IP Route Rules

A network device may only have one default network gateway address which receives transmitted IP packets that are not on the local subnet or do not match a custom routing rule. However, a single network gateway may not be sufficient or desirable in many network topologies. For example, LAN1 and LAN2 may each require their own network gateway for access. Figure 7 illustrates one such network topology.

Figure 7. Example of a Network Topology



Two routes are required in this topology so that:

1. LAN1 can access LAN1 Subnet B
2. LAN2 can access LAN2 Subnet B

The IP Router 1 may be configured as the default network gateway. However, the DMG2000 must also be configured to use IP Router 2 as a network gateway. A custom route must be created so that LAN2 can access LAN2 Subnet B. In addition, the default network gateway may be optionally replaced by another custom route. The custom route would allow LAN1 to access LAN1 Subnet B without using a default network gateway.

A default network gateway route blindly forwards IP packets whose destination is not the local subnet. A custom route is specific. It contains a single routing rule that specifically states when an IP packet is forwarded to a specific network gateway. This means that multiple custom routes may be installed on the same gateway to access multiple subnets. Each route contains a configured rule that directs the LAN interface, subnet mask and network gateway.

The advanced IP route rules include the following parameters:

- Ethernet Interface
- Destination Address
- Destination Mask
- Gateway Address

An IP route rule is a custom routing rule on a specific LAN interface. It allows access to a specific subnet that matches the destination address/mask via the specified gateway address. This rule supplements or replaces the functionality of the default network gateway address by providing access to multiple subnets across all LAN interfaces. The IP route rules are available on the DMG2000 only.

Ethernet Interface

Description: Specifies the LAN interface used for the route. The "Gateway Address" must be on the same subnet as this LAN interface.

Allowed Values:

- LAN1 = IP route will use LAN1
- LAN2 = IP route will use LAN2

Default Value = LAN1

INI File Parameter Name = ipRouteIf

Note: Unit requires a restart if this parameter value is changed.

Destination Address

Description: Destination IP address that is combined with the destination mask to define the subnet range for the route. Unused bits may be set to 0.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = blank

INI File Parameter Name = ipRouteDest

Note: Unit requires a restart if this parameter value is changed.

Destination Mask

Description: Destination IP mask that is combined with the destination address to define the subnet range for the route.

Allowed Values: Any valid IP mask

Default Value = blank

INI File Parameter Name = ipRouteMask

Note: Unit requires a restart if this parameter value is changed.

Gateway Address

Description: IP address of the gateway router that receives all IP packets from the LAN that match this route. The gateway IP address must be in the same subnet as the route's LAN interface.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = blank

INI File Parameter Name = ipRouteGw

Note: Unit requires a restart if this parameter value is changed.

3.2 Security Parameters

The Security Parameters support Secure Real-time Transport Protocol (SRTP) as well as SIP Transport Layer Security (TLS) and include the following groups:

- Certificate Management Group
- SRTP Management Group

3.2.1 Certificate Management Group

The Security parameters in the Certificate Management Group include:

- TLS Certificate Type
- HTTPS Certificate Type

3.2.1.1 TLS Certificate Type

Description: Determines whether the Media Gateway uses a self-assigned or Certificate Authority signed certificate for SIP Transport Layer Security (TLS).

Allowed Values:

- Self Signed = Use the self-signed TLS certificate
- CA Signed = Use the Certificate Authority-signed TLS certificate

Default Value = Self Signed

INI File Parameter Name = secSipTlsUseSelfSignedCert

Note: Unit requires a restart if this parameter value is changed.

3.2.1.2 HTTPS Certificate Type

Description: Determines whether the Media Gateway uses a self-assigned or CA signed certificate for HTTP over TLS (HTTPS).

Allowed Values:

- Self Signed = Use the self-signed TLS certificate
- CA Signed = Use the CA-signed TLS certificate

Default Value = Self Signed

INI File Parameter Name = secSipTlsUseSelfSignedCert

Note: Unit requires a restart if this parameter value is changed.

3.2.2 SRTP Management Group

The Secure Real-time Transport Protocol (RTP) group includes the following parameters:

- SRTP Preference
- MKI on Transmit Stream
- Key Derivation Enable
- Key Derivation Rate
- Anti-replay window size hint
- Cipher Mode
- Authentication Type
- Authentication Tag Length

3.2.2.1 SRTP Preference

Description: Specifies whether the Media Gateway and server exchange voice packets as encrypted or non-encrypted.

Allowed Values:

- RTP_Only = The Media Gateway uses only RTP. If the server only supports Secure RTP, then the calls are rejected.
- SRTP_Only = The Media Gateway uses only Secure RTP. If the server does not support Secure RTP, then the calls are rejected.
- SRTP_Preferred = The Media Gateway attempts to connect using Secure RTP. If the server supports Secure RTP, the call is connected using Secure RTP. If the server does not support Secure RTP, then the call is connected using RTP.

  Note: SRTP_Preferred is not supported in Version 5.1 SU2 Software.

Default Value = RTP_Only

INI File Parameter Name = srtpPreference

Note: Unit requires a restart if this parameter value is changed.

3.2.2.2 MKI on Transmit Stream

Description: Specifies if Master Key Index (MKI) is supported on transmit stream.

Allowed Values:

- Yes = Transmit stream supports MKI
- No = Transmit stream does not support MKI

Default Value = Yes

INI File Parameter Name = srtpTxMkiEnable

Note: Currently, the Media Gateway uses only one master key.

3.2.2.3 Key Derivation Enable

Description: Specifies if the Secure RTP (SRTP) Key is changed during a voice session.

Allowed Values:

- Yes = The SRTP Key is periodically changed during a voice session. The period is determined by the Key Derivation Rate.
- No = A single SRTP Key is used during a voice session.

Default Value = Yes

INI File Parameter Name = srtpKdrEnable

3.2.2.4 Key Derivation Rate

Description: Specifies the number of voice packets that cause the SRTP Key to be changed. The number is 2**<KDR>.

Allowed Values: 16 to 24

Default Value = 16

INI File Parameter Name = srtpKdrValue

3.2.2.5 Anti-replay window size hint

Description: Specifies the Anti-replay window size hint.

Allowed Values: 64 to 99

Default Value = 64

INI File Parameter Name = srtpWsh

3.2.2.6 Cipher Mode

Description: Specifies the cipher used to encrypt voice packets.

Note: Plain text is essentially no encryption and should be used only for testing purposes. For no encryption, you should specify RTP ONLY in the SRTP Preference parameter.

Allowed Values:

- Plain_Text = Use Plain_Text as the cipher (No encryption - for testing purposes only)
  Note: Plain_Text is not supported in Version 5.1 SU2 Software.
- AES_Counter_Mode = Use AES_Counter_Mode as the cipher

Default Value = AES_Counter_Mode

INI File Parameter Name = srtpEncodeType

3.2.2.7 Authentication Type

Description: Specifies the type of packet authentication used with Secure RTP. It is recommended that you enable authentication when using Secure RTP.

Allowed Values:

- None = No authentication is used
  Note: None is not supported in Version 5.1 SU2 Software.
- SHA1 = SHA1 authentication type is used

Default Value = SHA1

INI File Parameter Name = srtpAuthType

3.2.2.8 Authentication Tag Length

Description: Specifies the length of the authentication tag transmitted with the voice packet. A 32-bit tag length should be used when network loading is a concern.

Allowed Values:

- SHA1_32_bit = Uses 32-bit authentication tag
- SHA1_80_bit = Uses 80-bit authentication tag

Default Value = SHA1_80_bit

INI File Parameter Name = srtpAuthTag

3.3 System Parameters

These parameters control the system level operation of the Media Gateway. The System parameters include the following groups:

- System Group
- Telephony Group
- Serial Port, COM1 Group
- Serial Port, COM2 Group (DMG2000 Only)

3.3.1 System Group

The following parameter is in the System Group:

3.3.1.1 Operating Mode

Description: Selects the VoIP protocol (SIP or H.323) of the Media Gateway.

Allowed Values:

- H.323 = H.323 protocol is used.
- SIP = SIP protocol is used.

Default Value = SIP

INI File Parameter Name = sysOperatingMode

Note: Unit requires a restart if this parameter value is changed.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.3.2 Telephony Group

The System parameters in the Telephony group include:

- PBX Type
- PCM Coding

3.3.2.1 PBX Type

Description: Specifies the type of PBX telephony switch to which the DMG1000 is connected.

Note: This parameter does not apply to the Models DMG1008MTLDNI, DMG1008LS, DMG1004LS, DMG2030DTI, DMG2060DTI, and DMG2120DTI.

Note: The list of allowed values that will be displayed depends on your model type.

Allowed Values:

- None = No PBX Type has been selected.

- M1 = Nortel Meridian-1/Meridian SL-1
- Norstar = Nortel Norstar Key Systems
- Optiset_300ECS = Siemens Hicom 300E CS
- Optiset_300E = Siemens Hicom 300E
- Lucent = Lucent DEFINITY G3
- Magix = Merlin Magix
- NEC_IMG = NEC IMG
- NEC_IMX = NEC IMX
- NEC_NEAX = NEC NEAX or NEC 2400 IPX
- Rolm_8000 = Rolm 8000
- Rolm_9751_SW9005 = Rolm 9751 with software release 9005 or earlier
- Rolm_9751_SW9006 = Rolm 9751 with software release 9006 or earlier

Default Value = None

INI File Parameter Name = telPbxType

Note: If Optiset_300E is selected as the Telephony Switch Type, the setting of subscriber Message Waiting Indicator (MWI) lights is not supported and the Analog version of the DMG1000 (Models DMG1008LS, DMG1004LS) should be used to set/clear subscriber MWIs on this switch type.

Note: Unit requires a restart if this parameter value is changed.

3.3.2.2 PCM Coding

Description: Sets the Media Gateway to the PCM coding mode used by the PBX.

Allowed Values:

- µLaw = PBX uses µLaw coding.
- aLaw = PBX uses ALaw coding.

Default Value = µLaw

INI File Parameter Name = dspPcmCoding

3.3.3 Serial Port, COM1 Group

The Serial Port, COM1 group parameters are used to configure the DMG1000 serial interface port DIAGNOSTICS connector and the DMG2000 COM1 port connector. The system parameters in the Serial Port group include:

- Serial Port Baud Rate
- Serial Port Parity
- Serial Port Data Bits
- Serial Port Stop Bits

3.3.3.1 Serial Port Baud Rate

Description: Specifies the baud rate of the Media Gateway serial port.

Allowed Values:

- 1200 = 1200 bps
- 2400 = 2400 bps
- 9600 = 9600 bps
- 19200 = 19200 bps
- 38400 = 38400 bps

Default Value = 38400

INI File Parameter Name = sysSerialBaudRate

3.3.3.2 Serial Port Parity

Description: Specifies the parity of the Media Gateway serial port.

Allowed Values:

- None = Parity will not be used.
- Even = Even parity will be used.
- Odd = Odd parity will be used.

Default Value = None

INI File Parameter Name = sysSerialParity

3.3.3.3 Serial Port Data Bits

Description: Specifies the number of data bits used by the Media Gateway serial port.

Allowed Values:

- 7 data bits
- 8 data bits

Default Value = 8 data bits

INI File Parameter Name = sysSerialDataBits

3.3.3.4 Serial Port Stop Bits

Description: Specifies the number of stop bits used by the Media Gateway serial port.

Allowed Values:

- 1 stop bit
- 2 stop bits

Default Value = 1 stop bit

INI File Parameter Name = sysSerialStopBits

3.3.4 Serial Port, COM2 Group (DMG2000 Only)

The Serial Port, COM2 group parameters are used to configure the DMG2000 diagnostics/administration serial port (COM2 connector). The system parameters in the Serial Port group include:

- Serial Port Baud Rate
- Serial Port Parity
- Serial Port Data Bits
- Serial Port Stop Bits

3.3.4.1 Serial Port Baud Rate

Description: Specifies the baud rate of the DMG2000 serial port.

Allowed Values:

- 1200 = 1200 bps
- 2400 = 2400 bps
- 9600 = 9600 bps
- 19200 = 19200 bps
- 38400 = 38400 bps
- 57600 = 57600 bps
- 115200 = 115200 bps

Default Value = 115200 bps

INI File Parameter Name = sysSerialBaudRateCom2

3.3.4.2 Serial Port Parity

Description: Specifies the parity of the DMG2000 serial port.

Allowed Values:

- None = Parity will not be used.
- Even = Even parity will be used.
- Odd = Odd parity will be used.

Default Value = None

INI File Parameter Name = sysSerialParityCom2

3.3.4.3 Serial Port Data Bits

Description: Specifies the number of data bits used by the DMG2000 serial port.

Allowed Values:

- 7 data bits
- 8 data bits

Default Value = 8 data bits

INI File Parameter Name = sysSerialDataBitsCom2

3.3.4.4 Serial Port Stop Bits

Description: Specifies the number of stop bits used by the DMG2000 serial port.

Allowed Values:

- 1 stop bit
- 2 stop bits

Default Value = 1 stop bit

INI File Parameter Name = sysSerialStopBitsCom2

3.4 Analog Parameters (Models DMG1008LS, DMG1004LS Only)

The Analog parameters only apply to the Model DMG1008LS, DMG1004LS product. The Analog parameters include the following groups:

- Timing Group
- Feature Codes Group
- Message Waiting Control Group
- CPID Group

3.4.1 Timing Group

The Analog parameters in the Timing group include:

- Flash Hook
- Loop Current Off Debounce

- Incoming Rings Before Answer
- Ringing Timeout

3.4.1.1 Flash Hook

Description: Specifies the duration in milliseconds that the DMG1000 will remain on-hook during a hook flash operation.

Allowed Values: 50 to 4000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = telFlashMs

3.4.1.2 Loop Current Off Debounce

Description: Specifies the time in milliseconds that loop current can be removed by the PBX on an active call before the DMG1000 considers the line disconnected.

Allowed Values: 500 to 10000 milliseconds

Default Value = 2500 milliseconds

INI File Parameter Name = telLCOff

3.4.1.3 Incoming Rings Before Answer

Description: Specifies the number of ring bursts that must be present at the DMG1000 before the call is presented to the IP destination as a new call.

Allowed Values: 1 to 100

Default Value = 1

INI File Parameter Name = telIncomRing

3.4.1.4 Ringing Timeout

Description: Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away.

Allowed Values: 100 to 10000 milliseconds

Default Value = 6000 milliseconds

INI File Parameter Name = telRingOffMs

3.4.2 Feature Codes Group

The Analog parameters in the Feature Codes group include:

- Consult Call Dialtone Drop Code
- Consult Call Proceeding Drop Code
- Consult Call Busy Drop Code
- Consult Call Connected Drop Code
- Consult Call Disconnected Drop Code
- Consult Call Error Drop Code

3.4.2.1 Consult Call Dialtone Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Dialtone state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFaxCDropDt

3.4.2.2 Consult Call Proceeding Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Proceeding state (dialed but not connected) and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropProc

3.4.2.3 Consult Call Busy Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Busy state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropBusy

3.4.2.4 Consult Call Connected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Connected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropCon

3.4.2.5 Consult Call Disconnected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Disconnected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDis

3.4.2.6 Consult Call Error Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Error state and to reconnect to the original call.

Allowed Values: Dialable number, including '!' for flash hook and 'p' for pause.

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropError

3.4.3 Message Waiting Control Group

The Analog parameters in the Message Waiting Control group include:

- MWI Confirmation Tone
- Use Same Port for MWI Clear/Set

3.4.3.1 MWI Confirmation Tone

Description: Specifies whether or not the PBX sends a confirmation tone to signal the successful completion of a Message Waiting Indication (MWI) request.

Allowed Values:

- Yes = PBX will send a confirmation tone.
- No = PBX will not send a confirmation tone.

Default Value = No

INI File Parameter Name = telMwiConfirm

3.4.3.2 Use Same Port for MWI Clear/Set

Description: Specifies if the PBX requires that the telephony station that set an MWI be used to clear the MWI.

Note: Some PBX types require that the port that set an MWI be the same port that clears the MWI.

Allowed Values:

- Yes = Use the same port to set and clear the MWI.
- No = Not necessary to use the same port to clear an MWI that was used to set the MWI.

Default Value = Yes

INI File Parameter Name = telMwiSamePort

3.4.4 CPID Group

The Analog parameters in the Inband CPID group include:

- Initial Wait for Inband CPID
- Inband CPID Complete Timeout
- Analog Interface Type
- Central Office (Type I) Caller ID Type
- Central Office (Type I) Caller ID Alert Type
- Central Office (Type I) FSK Caller ID Expiration
- Central Office (Type I) FSK Caller ID Timeout

- CPID Parsing Configuration

3.4.4.1 Initial Wait for Inband CPID

Description: Specifies the number of milliseconds the DMG1000 will wait for the first inband DTMF digit to arrive after answering an incoming PBX call.

Allowed Values: 100 to 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telInbCpidStartMs

3.4.4.2 Inband CPID Complete Timeout

Description: Specifies the number of milliseconds the DMG1000 will wait for each subsequent inband DTMF digit to arrive. If no digit is received within this time, the CPID is assumed to be complete.

Allowed Values: 100 to 2000 milliseconds

Default Value = 300 milliseconds

INI File Parameter Name = telInbCpidEndMs

3.4.4.3 Analog Interface Type

Description: Specifies the type of analog interface to which the DMG1000 is connected.

Allowed Values:

- PBX
- Central Office

Default Value = PBX

INI File Parameter Name = telAlgIfType

3.4.4.4 Central Office (Type I) Caller ID Type

Description: Specifies the Type I Caller ID modulation to use, either Bellcore FSK (North America and Australia), DTMF (parts of Europe, including Sweden, Finland, Denmark, Netherlands, and Iceland), or none. This configuration option is only enabled when the Analog Interface Type is Central Office.

Note: If the Caller ID type is DTMF, then CPID parsing rules will need to be specified. See Chapter 6, "Media Gateway Parsers" for further information on the creation and use of parsing rules.

Allowed Values:

- Bellcore FSK
- DTMF
- None

Default Value = None

INI File Parameter Name = telAlgCoCidType

3.4.4.5 Central Office (Type I) Caller ID Alert Type

Description: Specifies the kind of Type I caller ID alert that is used on the trunk(s) connected to the DMG1000, either pause in ring cycle, ring burst, polarity reversal, or none. This configuration option is only enabled when the Central Office (Type I) Caller ID Type is set to **Bellcore FSK**.

Notes: ***1.*** If the pause in ring cycle alert type is used, the Incoming Rings Before Answer parameter should be set to greater than 1 for the caller ID information to be contained in the outgoing voice over IP (VoIP) message.

2. If an alert is used on the trunk that is not explicitly supported in the list (for example, OSI or a dual tone), use the None setting for this parameter.

Allowed Values:

- Pause in Ring Cycle
- Ring Burst
- Polarity Reversal
- None

Default Value = Pause in Ring Cycle

INI File Parameter Name = telAlgCoCidAlertType

3.4.4.6 Central Office (Type I) FSK Caller ID Expiration

Description: Specifies the duration in milliseconds that FSK Type I Caller ID information stays valid. This configuration option is only enabled when the Central Office Caller ID Type is Bellcore FSK.

Allowed Values: 100 to 60000 milliseconds

Default Value = 10000 milliseconds

INI File Parameter Name = telAlgTypeICidExpMs

3.4.4.7 Central Office (Type I) FSK Caller ID Timeout

Description: Specifies the duration in milliseconds that the digital signal processor (DSP) will perform FSK Type I Caller ID detection after an alert is detected. This configuration option is only enabled when the Central Office Caller ID Type is Bellcore FSK and the Caller ID Alert Type is other than None.

Allowed Values: 500 to 30000 milliseconds

Default Value = 5000 milliseconds

INI File Parameter Name = telAlgFskCidTimeoutMs

3.4.4.8 CPID Parsing Configuration

Description: Modifying this data affects either the in-band Type I (on-hook) or Type II (off-hook) CPID parsing of the DMG1000, depending on how the unit is configured. The text box on this page can be easily modified by either pasting parsing rules into it from a text file or directly modifying the contents of the control. See Chapter 6, "Media Gateway Parsers" for further information about the creation and use of parsing rules.

3.5 Serial Protocol Parameters

Note: The Serial Protocol Web page does not exist for Mitel or Rolm DMG1000 units.

The Serial Protocol parameters are used to define the serial protocol used in the connection to the PBX serial link. These parameters include:

- Serial Mode (Master/Slave)
- Serial Interface Protocol
- MCI Message Extension Length
- MCI Message Type
- CPID Length
- CPID Padding String
- Voice Mail Port Length
- System Number
- MWI Response Timeout
- IP Address of Serial Server
- Serial CPID Expiration
- Logical Extension Number

3.5.1 Serial Mode (Master/Slave)

Description: Specifies if the Media Gateway is using the serial protocol and, if so, whether the unit is the Serial Protocol Master or Slave. The Media Gateway must be configured as the Master if it is physically connected to the serial link of the PBX. In a system with multiple Media Gateways, only one of the units can be connected to the serial link of the PBX. This unit is the Serial Protocol Master and all other Media Gateways are considered to be the slaves.

Note: The master and slaves use the IP network to communicate serial protocol information to each other.

Allowed Values:

- None = Serial link is not used.
- Master = This Media Gateway is connected to the PBX serial link.
- Slave = This Media Gateway is part of a multiple configuration, but is not connected to the PBX serial link.

Default Value = None

INI File Parameter Name = telSerMode

Note: Unit requires a restart if this parameter value is changed.

3.5.2 Serial Interface Protocol

Description: Sets the Media Gateway to the serial protocol used by the PBX. Only valid when the Serial Mode parameter is set to Master.

Allowed Values:

- SMDI
- MCI
- MD110

Default Value = SMDI

INI File Parameter Name = telSerProtocol

Note: Unit requires a restart if this parameter value is changed.

3.5.3 MCI Message Extension Length

Description: Specifies the extension length used in MCI messages. Messages with six-digit extensions or with eight-digit extensions.

Allowed Values:

- Six-Digits
- Eight-Digits

Default Value = Six-Digits

INI File Parameter Name = telSerMciMsg

3.5.4 MCI Message Type

Description: Specifies the type of MCI messages. Messages can be Type B (default) or Type A. Type B messages include a tenant number of '01' while Type A messages exclude a tenant number.

Allowed Values:

- Type_A
- Type_B

Default Value = Type_B

INI File Parameter Name = telSerMciMsgType

3.5.5 CPID Length

Description: For SMDI, specifies the length of the extension field in an MWI request sent to the PBX from the DMG1000. For MD110, specifies the length of the calling and called party information contained in the CPID serial data packet from the PBX.

This parameter is only required when the serial protocol is set to SMDI or MD110 and the Media Gateway is the Serial Protocol Master.

Allowed Values: 2 to 10

Default Value = 7

INI File Parameter Name = telSerCpidLen

3.5.6 CPID Padding String

Description: Specifies the pad string to strip from the CPID fields (calling and called parties) in the incoming serial packet data. Also used to pad extensions in MWI requests sent to the PBX from the Media Gateway. This pad string must match the pad string configured by the PBX.

This parameter is only required when the Serial Protocol parameter is set to SMDI and the Media Gateway is the Serial Protocol Master.

Allowed Values: String of up to 10 digits

Default Value = (no default value)

INI File Parameter Name = telSerCpidPadStr

3.5.7 Voice Mail Port Length

Description: Specifies the length of the field in the serial data packet that contains the "voice mail port" number that the call arrived (or will arrive) on. This parameter is only required when the serial protocol is set to MD110 and the Media Gateway is the Serial Protocol Master.

Allowed Values: 2 to 5

Default Value = 2

INI File Parameter Name = telSerVmpLen

3.5.8 System Number

Description: Specifies the "voice mail system" that the serial protocol packet is being generated from. This parameter is used by the PBX in the event there is more than one application server (e.g. voice mail, unified messaging server, etc.) connected to the PBX. This parameter is only required when the serial protocol parameter is set to a value of MD110 and this Media Gateway is configured as the Serial Protocol Master.

Allowed Values: 0 to 99

Default Value = 1

INI File Parameter Name = telSerSysNum

3.5.9 MWI Response Timeout

Description: Specifies the time in milliseconds the Media Gateway serial protocol Master will wait for a failed response from the PBX before sending a success message to a serial protocol Slave in response to a message waiting indicator (MWI) request from the slave. This parameter is only required when this Media Gateway is configured as the Serial Protocol Master.

Allowed Values: 100 to 60000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telSerMwiRspToutMs

3.5.10 IP Address of Serial Server

Description: Specifies the IP Address of the Media Gateway that is configured as the Serial Protocol Master. Serial Protocol slave devices send and receive all serial protocol information to/from the Master device via the IP network. This parameter is only required when the Media Gateway is configured as the Serial Protocol Slave.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = telSerAddrSrvr

Note: Unit requires a restart if this parameter value is changed.

3.5.11 Serial CPID Expiration

Description: Specifies the time in milliseconds that serial CPID information received by a Media Gateway (Master or Slave) remains valid. If the timeout expires before an inbound call is received on the Media Gateway PBX port indicated by the serial CPID information, the serial CPID information is discarded.

Allowed Values: 100 to 60000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telSerCpidExpMs

3.5.12 Logical Extension Number

Description: Specifies the extension number assigned to each PBX port on the Media Gateway. This information must match the terminal numbers assigned to the ports connected to the Media Gateway in serial protocol administration at the PBX.

Allowed Values: 0 to 9999

Default Value = 0

INI File Parameter Name = telSerClientLtn

Note: Unit requires a restart if this parameter value is changed.

3.6 Gateway Parameters

The Gateway parameters include the following groups:

- Gateway Routing Group
- Gateway Advanced Group
- Gateway Capabilities Group

3.6.1 Gateway Routing Group

The Gateway Routing group includes the following subgroups:

- Call Routing Group

3.6.1.1 Call Routing Group

The Call Routing group includes the following parameters:

- Fault Tolerance Enabled
- Load Balancing Enabled
- VoIP Endpoint ID

Fault Tolerance Enabled

Description: Enables or disables fault-tolerance handling of outbound Voice over IP (VoIP) calls. If Yes is selected, the Media Gateway will failover to the next configured VoIP endpoint if an outbound VoIP call attempt fails. If No is selected, then a failed outbound VoIP call attempt will not be retried.

Allowed Values:

- Yes = Media Gateway will failover to next configured VoIP endpoint if a call attempt fails.
- No = Media Gateway failed call attempt will not be retried.

Default Value = No

INI File Parameter Name = gwFaultToleranceEnabled

Load Balancing Enabled

Description: Enables or disables load-balancing of outbound Voice over IP (VoIP) calls. If Yes is selected, then the Media Gateway will route outbound VoIP calls to each configured VoIP endpoint in a round-robin fashion. If No is selected, the Media Gateway will not load-balance across multiple VoIP endpoints.

Allowed Values:

- Yes = Media Gateway will perform load-balancing on outbound VoIP calls.
- No = Media Gateway will not perform load-balancing on outbound VoIP calls.

Default Value = No

INI File Parameter Name = gwLoadBalanceEnabled

VoIP Endpoint ID

Description: Specifies the IP address of the Voice over IP (VoIP) Server(s) that will receive all inbound calls from the PBX.

Note: Additional VoIP Servers can be added to the list by pressing the **Add New Endpoint** button. Each VoIP Endpoint entered will then be used as part of the Fault Tolerance and/or Load Balancing schemes as defined by the Fault Tolerance Enabled and Load Balancing Enabled parameters.

Allowed Values:

- If no H.323 Gatekeeper, SIP Registration Server, or DNS Server is used, then any valid IP address (in dotted decimal notation) may be specified.
- If an H.323 Gatekeeper, SIP Registration Server, or DNS Server is used, then the IP terminal address may also be specified as an alias string name, an e-mail address, or an IP address (in dotted decimal notation).

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

Default Value = None

INI File Parameter Name = gwVoIPEndPointId

3.6.2 Gateway Advanced Group

The Gateway Advanced group includes the following subgroups:

- Advanced Call Routing Group
- Telephony Group
- Audio Group
- Fax Group
- Quality of Service Group
- E-Mail Group
- SysLog Group
- SNMP Group
- Web Server Group
- Telnet Server Group

3.6.2.1 Advanced Call Routing Group

Note: The Advanced Call Routing subgroup Gateway parameters only apply to a Media Gateway operating in the Phone Emulating Mode.

The Advanced Call Routing subgroup includes the following parameters:

- Call Connect Mode
- Send DNIS to VoIP Endpoint
- Destination for Un-routable IP Calls
- Destination for Un-routable PBX Calls
- Monitor Call Connections
- Call Monitor Interval

Call Connect Mode

Description: Specifies when voice path is created between the IP device endpoint and the PBX on IP-to-PBX calls.

Allowed Values:

- On-Answer = the voice path is not established until the dialed PBX extension answers the call.
- Instant = the voice path is established as soon as the PBX extension is dialed.

Default Value = Instant

INI File Parameter Name = gwCallConnectMode

Send DNIS to VoIP Endpoint

Description: Specifies whether Dialed Number Identification Service (DNIS) received on inbound TDM call is passed to the VoIP endpoint.

Allowed Values:

- Yes = DNIS is passed to the VoIP endpoint.
- No = DNIS is not passed to the VoIP endpoint.

Default Value = No

INI File Parameter Name = gwSendDnisToVoip

Destination for Un-routable IP Calls

Description: Specifies the IP terminal address to which un-routable IP-To-PBX calls will be forwarded.

Allowed Values:

- If no H.323 Gatekeeper or SIP Registration Server is used, then any valid IP address may be specified.
- If there is an H.323 Gatekeeper or SIP Registration Server, then the IP terminal address may be specified as an alias string name, an e-mail address, or an IP address.
- If blank (none), then un-routable IP-To-PBX calls are rejected.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

Default Value = (no default value)

INI File Parameter Name = gwDefaultIPDest

Destination for Un-routable PBX Calls

Description: Specifies the PBX extension number to which un-routable PBX-To-IP calls will be transferred.

Allowed Values:

- Any valid dialable number string (0 to 7 characters).
- If blank (none), then un-routable calls are ignored/dropped.

Default Value = (no default value)

INI File Parameter Name = gwDefaultPBXCallDest

Monitor Call Connections

Description: When the Monitor Call Connections parameter is enabled (set to "Yes"), the Media Gateway will monitor the connection state of active IP calls. If the active IP call has lost connection, the Media Gateway will tear down the call.

Note: If enabled in SIP, the IP endpoints must support session timers. If enabled in H.323, the IP endpoints must support Real-Time-Delay requests.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

Allowed Values:

- Yes = the Media Gateway monitors the connection state of active IP calls.
- No = the Media Gateway does not monitor the connection state of active IP calls.

Default Value = No

INI File Parameter Name = gwMonitorCallConns

Call Monitor Interval

Description: Specifies the call monitor interval in seconds at which active IP calls are monitored in order to determine the connections status. This parameter is only valid if the Monitor Call Connections parameter is enabled.

Allowed Values: 30 through 3600 seconds

Default Value = 60

INI File Parameter Name = gwMonitorCallIntSec

3.6.2.2 Telephony Group

The Telephony subgroup includes the following parameters:

- Minimum Call Party Delay (Phone Emulating Only)

- Maximum Call Party Delay (Phone Emulating Only)
- Dial Digit On Time
- Dial Inter-Digit Time
- Dial Pause Time
- Turn MWI On FAC (Phone Emulating Only)
- Turn MWI Off FAC (Phone Emulating Only)
- Dial Send Key (Phone Emulating Only)
- Outbound Call Connect Timeout (Phone Emulating Only)
- Wait for Ringback/Connect on Blind Transfer (Phone Emulating Only)
- Hunt Group Extension (Phone Emulating Only)

Minimum Call Party Delay (Phone Emulating Only)

Description: Specifies the minimum number of milliseconds that the Media Gateway will wait for display information (calling party information) on an inbound PBX call before the call is routed to the IP destination. (0 for no delay)

Allowed Values: 0 through 10000 (msec)

Default Value = 0

INI File Parameter Name = telMinWaitForCpid

Maximum Call Party Delay (Phone Emulating Only)

Description: Specifies the maximum number of milliseconds that the Media Gateway will wait for display information (calling party information) on an inbound PBX call before the call is routed to the IP destination. (0 for no delay)

Allowed Values: 0 through 10000 (msec)

Default Value = 2000

INI File Parameter Name = telCallInfoTout

Dial Digit On Time

Description: Specifies the duration in milliseconds of a dialed DTMF digit to the PBX.

Allowed Values: 30 through 2000 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = telDialDigitOnMs

Dial Inter-Digit Time

Description: Specifies the delay in milliseconds between dialed DTMF digits to the PBX.

Allowed Values: 30 through 2000 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = telDialInterDigitMs

Dial Pause Time

Description: Specifies the delay in milliseconds for each pause (,) character encountered in a dial string.

Allowed Values: 40 through 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telDialPauseMs

Turn MWI On FAC (Phone Emulating Only)

Description: Specifies the Features Access Code (FAC) to dial to turn on the Message Waiting Indicator (MWI) of a PBX extension.

Allowed Values: Any string of less than 11 characters.

Default Value = (no default value)

INI File Parameter Name = telMwiOnFAC

Turn MWI Off FAC (Phone Emulating Only)

Description: Specifies the Features Access Code (FAC) to dial to turn off the Message Waiting Indicator (MWI) of a PBX extension.

Allowed Values: Any string of less than 11 characters.

Default Value = (no default value)

INI File Parameter Name = telMwiOffFAC

Dial Send Key (Phone Emulating Only)

Description: Specifies the 'hot key' used to indicate the end of a dialed extension. For example, if the Dial Send Key is configured for an asterisk (*), the DMG1000 will dial extension 3200 as 3200*. For PBXs that have multiple length dial plans, this allows the DMG1000 to specify that the dialing is complete.

Allowed Values: *, #, None

Default Value = None

INI File Parameter Name = dpInbVoipAltRoute

Outbound Call Connect Timeout (Phone Emulating Only)

Description: Specifies the time in milliseconds that the Media Gateway will wait for a connect event on an outbound circuit call. If no connect event is received within the time specified, the Media Gateway will automatically transition the call to the connected state.

Allowed Values: 0 to 60000 milliseconds

Default Value = 10000 milliseconds.

INI File Parameter Name = telConnectToutMs

Wait for Ringback/Connect on Blind Transfer (Phone Emulating Only)

Description: Specifies whether the gateway will wait for the detection of ringback tone or connection before completing a blind transfer of a circuit call. If Yes is selected, blind transfers will wait for a ringback tone or connect event (voice) before completing the transfer. If No is selected, a blind transfer will be completed as soon as the destination number is dialed. Valid only in OnAnswer connect mode.

Allowed Values: Yes, No

Default Value = Yes

INI File Parameter Name = telBlindXfrWaitConfirm

Hunt Group Extension (Phone Emulating Only)

Description: If the telephony ports are configured in a Hunt Group on the switch, then enter the Hunt Group extension number in this field. The gateway will use this field to ignore the Hunt Group extension when generating the call party information. This ensures that the Hunt Group extension number is not mistaken for a calling or called party number.

Allowed Values: Any valid dialable number string up to 7 characters

Default Value = (no default value)

INI File Parameter Name = telHuntGroupExtn

3.6.2.3 Audio Group

The Audio subgroup includes the following parameters:

- Audio Compression
- RTP Digit Relay Mode
- RTP Fax/Modem Tone Relay Mode
- RTP Source IP Address Validation
- RTP Source UDP Port Validation
- Signaling Digit Relay Mode
- Voice Activity Detection
- Codec/Frame Size/Frames Per Packet

Audio Compression

Description: Sets the audio coder/decoder to be used by the Media Gateway.

Allowed Values:

- G.711u/G.711a = G.711 uLaw preferred, G.711 aLaw secondary
- G.711u = G.711 uLaw only
- G.711a = G.711 aLaw only
- G.723.1 = G.723.1 only
- G.729AB = G.729AB only

Default Value = G.711u/G.711a

INI File Parameter Name = dspCompression

Note: Unit requires a restart if this parameter value is changed.

RTP Digit Relay Mode

Description: Selects the Realtime Transfer Protocol (RTP) method by which dual tone multi-frequency (DTMF) tones are transported between the Media Gateway and VoIP endpoints.

Note: The Inband-Tone method passes the digits as audio data in the RTP stream, and is only reliable when the parameter **Audio Compression** is set to **G.711 only**. The AVT method uses RFC 2833 RTP packets to pass the digits.

Allowed Values (Phone Emulating):

- None = The Media Gateway will not use RTP packets to transmit/receive tone information.
- RFC2833 = The (DTMF) tone information is sent between the Media Gateway and VoIP endpoints via RTP packets as defined by RFC 2833.

  Note: A payload type of 101 is used in the implementation of the RFC2833 method.

- Inband-Tone = The DTMF tones are coded into the regular audio packets (RTP voice packets) sent between the Media Gateway and VoIP endpoints.

Default Value = RFC2833

INI File Parameter Name = dspDigitRelay

RTP Fax/Modem Tone Relay Mode

Description: Selects the Realtime Transfer Protocol (RTP) method by which fax and modem tones are transported between the TDM network and VoIP network.

Allowed Values:

- RFC2833 = Fax and modem tones from the TDM network are sent as RFC2833 packets to the VoIP network. RFC2833 packets from the VoIP network are rendered to TDM.
- Inband-Tone = Fax and modem tones from the TDM network are passed through VoIP. Inband-Tone is only reliable when the Preferred Code parameter is set to G.711 only.

Default Value = RFC2833

INI File Parameter Name = dspFaxModemToneRelay

RTP Source IP Address Validation

Description: If set to On, then the source IP address of received RTP packets must match the IP address to which RTP is being sent. If the source IP address does not match, then the packet is discarded. If set to Off, then the source IP address of received RTP packets is not validated.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = gwRTPValidateSrcIp

Note: Unit requires a restart if this parameter value is changed.

RTP Source UDP Port Validation

Description: If set to On, then the source UDP port of received RTP packets must match the UDP port to which RTP is being sent. If the source UDP port does not match, then the packet is discarded. If set to Off, then the source UDP port of received RTP packets is not validated.

Allowed Values:

- On
- Off

Default Value = Off

INI File Parameter Name = gwRTPValidateSrcPort

Note: Unit requires a restart if this parameter value is changed.

Signaling Digit Relay Mode

Description: Specifies if out-of-band messages are used to transport DTMF tones between the Media Gateway and VoIP endpoints.

Note: This parameter should be set to Off when the Operating Mode parameter is set to either SIP Emulating or SIP Driving.

Allowed Values:

- On
- Off

Default Value = On

INI File Parameter Name = gwSigDigitRelay

Voice Activity Detection

Description: Enables the use of voice activity detection (VAD) to reduce the amount of audio data traffic. If VAD is enabled, the unit will stop the transmission of RTP audio data when no voice activity is detected on the telephony port. Transmission is continued when voice activity is detected. This drastically reduces the amount of audio traffic on the network.

Allowed Values:

- On = Voice activity detection is enabled.
- Off = Voice activity detection is disabled.

Default Value = On

INI File Parameter Name = dspVAD

Codec/Frame Size/Frames Per Packet

Description: Refers to the type of coder/decoder used for voice compression. The parameter values for each Codec type are defined in Table 1, “Coder/Decoder Parameters”, on page 77.

Note: (For DMG1000 only) If secure RTP (SRTP) is enabled, then you must select 30 msec frame sizes for G.711.

Table 1. Coder/Decoder Parameters

Codec Type	Frame Size in Bytes	Frames Per Packet
DMG1000		
G.711	Selectable: 10, 20, 30	1
G.723	10	Selectable: 1 through 4
G.729	20	Selectable: 1 through 8
DMG2000		
G.711	Selectable: 10, 20, 30	1
G.723	10	1 or 2
G.729	20	Selectable: 1 through 6

INI File Parameter Name = dspFrameSizeG711, dspFramesPerPktG723k, dspFramesPerPktG729

3.6.2.4 Fax Group

The Fax subgroup includes the following parameter:

- Fax-IP Transport Mode

Fax-IP Transport Mode

Description: Defines the method used by the Media Gateway to transport fax calls over IP.

Allowed Values:

- None = No special processing is performed for transporting fax.
- T.38 = T.38 fax protocol is used.
- G.711- Passthrough = Media Gateway will automatically switch the IP media stream to G.711 Passthrough.

Default Value = T.38

INI File Parameter Name = gwFaxTransportMode

Note: Unit requires a restart if this parameter value is changed.

3.6.2.5 Quality of Service Group

The Quality of Service subgroup includes the following parameters:

- Call Control QOS Byte
- RTP QOS Type

Call Control QOS Byte

Description: The Call Control QOS Byte parameter defines a decimal byte value that represents QOS bit flags. This parameter is used in each call control (SIP or H.323) data packet transmitted from the Media Gateway with a QOS byte code. Routers use this byte code to assign priority levels to packets. The QOS byte may be interpreted as either IPv4 TOS or DiffServ.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

IPv4 TOS Byte

Bits:
0-2 = Precedence (RFC 1122)
3-6 = Type of Service (RFC 1349/1455)
7 = Must be zero

PrecedenceBits (0-2):
111xxxxx (224) = Network Control
110xxxxx (192) = Internetwork Control
101xxxxx (160) = CRITIC/ECP
100xxxxx (128) = Flash Override
011xxxxx (96) = Flash
010xxxxx (64) = Immediate
001xxxxx (32 = Priority
000xxxxx (0) = Routine

Type of Service Bits (3-6):
xxx0000x (0) = Normal
xxx1000x (16) = Minimize delay
xxx0100x (8) = Maximize throughput
xxx0010x (4) = Maximize reliability
xxx0001x (2) = Minimize monetary cost
xxx1111x (30) = Maximize physical link security
xxx (32 = Priority
xxx (0) = Routine

Default Value = 0 (Routine/Normal)

DiffServ Codepoint Byte

Bits:
0-5 = Differentiated Services Codepoint (RFC 2474)
6-7 = Explicit Congestion Notification (RFC 2481)

Allowed Values: 0-255

Default Value = 0

INI File Parameter Name = gwQosCallControl

RTP QOS Type

Description: The RTP QOS Type parameter defines a decimal value that represents QOS bit flags. This parameter is used in each RTP data packet transmitted from the Media Gateway with a QOS byte code. Routers use this byte code to assign priority levels to packets. The QOS byte may be interpreted as either IPv4 TOS or DiffServ.

IPv4 TOS Byte

Bits:
0-2 = Precedence (RFC 1122)
3-6 = Type of Service (RFC 1349/1455)
7 = Must be zero

PrecedenceBits (0-2):
111xxxxx (224) = Network Control
110xxxxx (192) = Internetwork Control
101xxxxx (160) = CRITIC/ECP
100xxxxx (128) = Flash Override
011xxxxx (96) = Flash
010xxxxx (64) = Immediate
001xxxxx (32 = Priority
000xxxxx (0) = Routine

Type of Service Bits (3-6):
xxx0000x (0) = Normal
xxx1000x (16) = Minimize delay
xxx0100x (8) = Maximize throughput
xxx0010x (4) = Maximize reliability
xxx0001x (2) = Minimize monetary cost
xxx1111x (30) = Maximize physical link security
xxx (32 = Priority
xxx (0) = Routine

Default Value = 0 (Routine/Normal)

DiffServ Codepoint Byte

Bits:
0-5 = Differentiated Services Codepoint (RFC 2474)
6-7 = Explicit Congestion Notification (RFC 2481)

Allowed Values: 0-255

Default Value = 0

INI File Parameter Name = gwQosRtp

3.6.2.6 E-Mail Group

The E-Mail subgroup includes the following parameters:

- E-Mail Alarms Enabled
- E-Mail Minimum Alarm Severity
- Destination E-Mail List
- E-Mail Server IP Address
- Source E-Mail Address

E-Mail Alarms Enabled

Description: Defines whether an e-mail is generated to signal an alarm.

Allowed Values:

- Yes = E-mail is generated.
- No = E-mail is not generated.

Default Value = No

INI File Parameter Name = emailEnabled

E-Mail Minimum Alarm Severity

Description: Defines the minimum alarm severity level that generates an e-mail notification.

Allowed Values:

- Error
- Warning
- Info

Default Value = Info

INI File Parameter Name = emailAlarmSeverity

Destination E-Mail List

Description: Semi-colon delimited list of e-mail addresses that will receive e-mail alarms.

Allowed Values: Semi-colon delimited list of e-mail addresses.

Default Value = (no default value)

INI File Parameter Name = emailList

E-Mail Server IP Address

Description: IP addresses of the SMTP e-mail server that receive the e-mail generated by the IP gateway.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = emailServerAddr

Source E-Mail Address

Description: E-mail address used as source of e-mail alarm messages.

Allowed Values: Any valid e-mail address

Default Value = alarm@pbxgw.com

INI File Parameter Name = emailSource

3.6.2.7 SysLog Group

The SysLog subgroup includes the following parameters:

- SysLog Server IP Address
- Alarms to SysLog Enabled
- SysLog Minimum Alarm Severity
- Diagnostics Trace to SysLog Enabled

SysLog Server IP Address

Description: IP address of the SysLog (RFC3164) server (notifications sent to server's UDP port 514).

Allowed Values: Any valid IP address in dotted decimal format

Default Value = (no default value)

INI File Parameter Name = syslogServerAddr

Note: Unit requires a restart if this parameter value is changed.

Alarms to SysLog Enabled

Description: Defines whether alarms are sent to the SysLog server.

Allowed Values:

- Yes = Alarms are sent as user-level facility messages.
- No = Alarms are sent to SysLog server.

Default Value = No

INI File Parameter Name = syslogAlarmEnabled

SysLog Minimum Alarm Severity

Description: Defines the minimum alarm severity level that generates notifications to SysLog server.

Allowed Values:

- Error = Error alarms are sent to the SysLog server.
- Warn = Error and Warning alarms are sent to the SysLog server.
- Info = Error, Warning, and Informational alarms are sent to the SysLog server.

Default Value = No

INI File Parameter Name = syslogAlarmSeverity

Diagnostics Trace to SysLog Enabled

Description: Defines whether diagnostic trace messages are sent to SysLog server.

Allowed Values:

- Yes = Diagnostics trace messages are sent as user-level facility/debug messages.
- No = Diagnostics trace messages are not sent as user-level facility/debug messages.

Default Value = No

INI File Parameter Name = syslogTraceEnabled

3.6.2.8 SNMP Group

The SNMP subgroup includes the following parameters:

- SNMP Traps Enabled?
- SNMP Minimum Alarm Severity
- SNMP Trap IP List
- SNMP Community Name
- SNMP System Name
- SNMP System Contact
- SNMP System Location

SNMP Traps Enabled?

Description: Defines whether an SNMP trap is generated to signal an alarm.

Allowed Values:

- Yes = SNMP trap is generated.
- No = SNMP trap is not generated.

Default Value = No

INI File Parameter Name = snmpTrapEnabled

SNMP Minimum Alarm Severity

Description: Defines the minimum alarm severity level to generate an SNMP trap.

Allowed Values:

- Error
- Warning
- Info

Default Value = Info

INI File Parameter Name = snmpAlarmSeverity

SNMP Trap IP List

Description: Semi-colon delimited list of the IP addresses of SNMP Managers who are to receive SNMP traps generated by the IP gateway.

Allowed Values: Semi-colon delimited list of IP addresses.

Default Value = 255.255.255.255

INI File Parameter Name = snmpTrapAddressList

SNMP Community Name

Description: Specifies the SNMP Community name. The specified community has read-only capabilities.

Allowed Values: Any string with length between 5 - 14 characters.

Default Value = public

INI File Parameter Name = snmpCommunity

Note: Unit requires a restart if this parameter value is changed.

SNMP System Name

Description: Specifies the SNMP System name.

Allowed Values: Any string with length between 0 - 63 characters.

Default Value = (no default value)

INI File Parameter Name = snmpSysName

Note: Unit requires a restart if this parameter value is changed.

SNMP System Contact

Description: Specifies the SNMP System administration contact name.

Allowed Values: Any string with length between 0 - 63 characters.

Default Value = (no default value)

INI File Parameter Name = snmpSysContact

Note: Unit requires a restart if this parameter value is changed.

SNMP System Location

Description: Specifies the SNMP System location.

Allowed Values: Any string with length between 0 - 63 characters.

Default Value = (no default value)

INI File Parameter Name = snmpSysLocation

Note: Unit requires a restart if this parameter value is changed.

3.6.2.9 Web Server Group

The Web Server subgroup includes the following parameters:

- HTTP Server Enabled
- HTTPs Server Enabled

HTTP Server Enabled

Description: Defines whether the HTTP Server is enabled or disabled. Disable the HTTP Server to disable non-secure Web access.

Allowed Values:

- Yes = HTTP Server is enabled.
- No = HTTP Server is disabled.

Default Value = Yes

INI File Parameter Name = webHttpEnabled

Note: Unit requires a restart if this parameter value is changed.

HTTPs Server Enabled

Description: Defines whether the HTTPs Server is enabled or disabled. Disable the HTTP Server to disable non-secure Web access.

Allowed Values:

- Yes = HTTPs Server is enabled.
- No = HTTPs Server is disabled.

Default Value = No

INI File Parameter Name = webHttpsEnabled

Note: Unit requires a restart if this parameter value is changed.

3.6.2.10 Telnet Server Group

The Telnet Server subgroup includes the following parameter:

- Telnet Server Enabled

Telnet Server Enabled

Description: Defines whether the Telnet Server is enabled. Disable the Telenet Server to prevent any Telenet connections to the Media Gateway.

Allowed Values:

- Yes
- No

Default Value = Yes

INI File Parameter Name = telnetEnabled

Note: Unit requires a restart if this parameter value is changed.

3.6.3 Gateway Capabilities Group

The Port Capabilities parameters allow you to individually configure the call control capabilities of each port connected to the Media Gateway. These parameters also allow you to enable or disable individual ports. The Port Capabilities group includes the following parameters:

- Port #
- Telephony Port Capability
- Telephony Port Enabled

3.6.3.1 Port

Description: This is a read-only parameter that defines the Media Gateway port numbers.

Note: If the T1/E1 signaling mode is configured for ISDN, the Port # refers to the T1 or E1 span number.

3.6.3.2 Telephony Port Capability

Description: Specifies the types of requests that the Media Gateway port may service.

Allowed Values:

- MWIs-Only = Port will only service MWI requests. The port should not be part of a PBX call hunt group.
- Calls-Only = Port will only service calls. MWI requests will not be serviced by this port.
- Both = Port will service both MWI requests and calls.

Default Value = Both

INI File Parameter Name = telPortCapability

3.6.3.3 Telephony Port Enabled

Description: Specifies whether the telephony port is disabled or enabled. Disabled ports drop communications links to the PBX.

Note: If the T1/E1 signaling mode is configured for ISDN, the Port # refers to the T1 or E1 span number.

Allowed Values:

- Yes = Port is enabled and is capable of providing a connection. The LED for this port may or may not be green, depending on whether or not a line is plugged in.
- No = Port is disabled and is incapable of providing a connection. The LED for this port will be red, regardless of whether or not a line is plugged in.

Default Value = Yes

INI File Parameter Name = telPortEnabled

3.7 T1/E1 Parameters (DMG2000 Only)

The T1/E1 parameters are used to configure the T1 and E1 port(s) on a DMG2000 and include the following groups:

- T1/E1 Mode Group
- T1 CAS Protocol Group (T1 CAS Signaling Mode)
- T1 ISDN Protocol Group (ISDN Signaling Mode)
- E1 ISDN Protocol Group (ISDN Signaling Mode)
- CPID Parsing Configuration

3.7.1 T1/E1 Mode Group

The parameters in the T1/E1 Mode group include the following:

- Line Mode
- Signaling Mode
- Interface Side

3.7.1.1 Line Mode

Description: Specifies the Line Mode type (T1 or E1) to which the T1/E1 connector will be interfacing.

Allowed Values:

- T1 = Interfaces to T1 line (23/24 channels)
- E1 = Interfaces to E1 line (30 channels)

Default Value = T1

INI File Parameter Name = t1e1LineMode

3.7.1.2 Signaling Mode

Description: Specifies the signaling mode to be used.

Allowed Values:

- CAS = Channel associated signaling
- ISDN = Common channel signaling

Default Value = CAS

INI File Parameter Name = t1e1Signaling

Note: Unit requires a restart if this parameter value is changed.

3.7.1.3 Interface Side

Description: Identifies the side of the connection.

Allowed Values:

- Terminal = Terminal (user) side of device
- Network = Network side of device

Default Value = Terminal

INI File Parameter Name = t1e1Interface

3.7.2 T1 CAS Protocol Group (T1 CAS Signaling Mode)

The parameters in the T1 CAS Protocol group include the following:

- T1 CAS Protocol
- Line Encoding
- Framing
- Selects Transmit Pulse Waveform
- Flash Hook
- Incoming Rings Before Answer
- Ringing Timeout (msec)
- Ring Cycle Time
- Consult Call Dialtone Drop Code
- Consult Call Proceeding Drop Code
- Consult Call Busy Drop Code
- Consult Call Connected Drop Code
- Consult Call Disconnected Drop Code
- Consult Call Error Drop Code
- MWI Confirmation Tone
- CPID Type
- Initial Wait for Inband CPID (msecs)
- Inband CPID Complete Timeout (msecs)

3.7.2.1 T1 CAS Protocol

Description: If T1 CAS is selected as the Signaling Mode, specifies the T1 CAS protocol to be used.

Allowed Values:

- Loop Start = Loop Start protocol will be used.

- Ground Start = Ground Start protocol will be used.
- E&M Immediate = E&M Immediate protocol will be used.
- E&M Delay = E&M Delay protocol will be used.
- E&M Wink = E&M Wink protocol will be used.

Default Value = Loop Start

INI File Parameter Name = t1e1CASProtocol

3.7.2.2 Line Encoding

Description: Specifies the type of T1 line coding that will be used.

Allowed Values:

- AMI = Alternate Mark Inversion line coding is used.
- B8ZS = Binary Eight Zero Substitution line coding is used.

Default Value = B8ZS

INI File Parameter Name = t1e1Encoding

3.7.2.3 Framing

Description: Specifies the type of T1 framing that will be used by the line.

Allowed Values:

- SF = D4 Superframe format (12 consecutive T1 frames) is used.
- ESF = Extended Superframe format (24 consecutive T1 frames) is used.

Default Value = ESF

INI File Parameter Name = t1e1Framing

3.7.2.4 Selects Transmit Pulse Waveform

Description: Selects the method to be used for generating the Transmit Waveform Shape.

Allowed Values:

- Short Haul 110 ft = The DMG2000 is 0 - 110 feet from T1 line source.
- Short Haul 220 ft = The DMG2000 is 110 - 220 feet from T1 line source.
- Short Haul 330 ft = The DMG2000 is 220 - 330 feet from T1 line source.
- Short Haul 440 ft = The DMG2000 is 330 - 440 feet from T1 line source.
- Short Haul 550 ft = The DMG2000 is 440 - 550 feet from T1 line source.
- Short Haul 660 ft = The DMG2000 is 550 - 660 feet from T1 line source.

Default Value = Short Haul 110 ft

INI File Parameter Name = t1e1TxWave

3.7.2.5 Flash Hook

Description: - Specifies the duration in milliseconds that the Media Gateway will remain on-hook during a hook flash operation.

Allowed Values: 50 to 4000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = telFlashMs

3.7.2.6 Incoming Rings Before Answer

Description: - Specifies the number of ring bursts that must be present at the DMG2000 before the call is presented to the IP destination as a new call.

Allowed Values: 1 to 100

Default Value = 1

INI File Parameter Name = telIncomRing

3.7.2.7 Ringing Timeout (msec)

Description: - Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away.

Allowed Values: 100 to 10000 milliseconds

Default Value = 6000 milliseconds

INI File Parameter Name = telRingOffMs

3.7.2.8 Ring Cycle Time

Description: Specifies the number of milliseconds of non-ringing that will signal that an incoming call has gone away[Duplicate of Ringing Timeout parameter. Will be obsoleted.]

Allowed Values: Numerical value from 1000 to 10000

Default Value = 6000

INI File Parameter Name = msRingCycleTime

3.7.2.9 Consult Call Dialtone Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Dialtone state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFaxCDropDt

3.7.2.10 Consult Call Proceeding Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Proceeding state (dialed but not connected) and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropProc

3.7.2.11 Consult Call Busy Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Busy state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropBusy

3.7.2.12 Consult Call Connected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Connected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropCon

3.7.2.13 Consult Call Disconnected Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Disconnected state and to reconnect to the original call.

Allowed Values: A string from 0 to 10 digits in length using any of the following characters: 0-9, *, #, !

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropDis

3.7.2.14 Consult Call Error Drop Code

Description: Defines the Feature Code to dial in order to drop a Consult call that is in the Error state and to reconnect to the original call.

Allowed Values: Dialable number, including '!' for flash hook and 'p' for pause.

Default Value = !!

Note: Each ! is a hook flash. For !, the Media Gateway will perform a single hook-flash operation. For !!, the Media Gateway will perform a double hook-flash operation.

INI File Parameter Name = telFacCDropError

3.7.2.15 MWI Confirmation Tone

Description: Specifies whether or not the PBX sends a confirmation tone to signal the successful completion of a Message Waiting Indication (MWI) request.

Allowed Values:

- Yes = PBX will send a confirmation tone.
- No = PBX will not send a confirmation tone.

Default Value = No

INI File Parameter Name = telMwiConfirm

3.7.2.16 CPID Type

Description: Selects the timing of in-band CPID information.

Allowed Values:

- Type I DTMF = CPID sent between first and second ring
- Type II DTMF = CPID sent after off-hook

Default Value = Type I DTMF

INI File Parameter Name = t1e1CpidType

3.7.2.17 Initial Wait for Inband CPID (msecs)

Description: Specifies the number of milliseconds the DMG2000 will wait for the first inband DTMF digit to arrive after answering an incoming PBX call.

Allowed Values: 100 to 10000 milliseconds

Default Value = 2000 milliseconds

INI File Parameter Name = telInbCpidStartMs

3.7.2.18 Inband CPID Complete Timeout (msecs)

Description: Specifies the number of milliseconds the DMG2000 will wait for each subsequent inband DTMF digit to arrive. If no digit is received within this time, the CPID is assumed to be complete.

Allowed Values: 100 to 2000 milliseconds

Default Value = 300 milliseconds

INI File Parameter Name = telInbCpidEndMs

3.7.3 T1 ISDN Protocol Group (ISDN Signaling Mode)

The parameters in the T1 ISDN Protocol group include the following:

- ISDN Protocol
- ISDN Protocol Variant
- Line Encoding
- Framing
- Selects Transmit Pulse Waveform

- Outbound TDM Calling Party Source
- Static TDM Calling Party

3.7.3.1 ISDN Protocol

Description: If ISDN is selected as the Signaling Mode, specifies the T1 ISDN protocol to be used.

Allowed Values:

- QSIG = QSIG ISDN protocol is used.
- NI-2 = US National ISDN - Phase 2 protocol is used.
- 5ESS = 5ESS protocol is used.
- DMS100 = DMS100 protocol is used.

Default Value = QSIG

INI File Parameter Name = t1IsdnProtocol

3.7.3.2 ISDN Protocol Variant

Description: Specifies the type of T1 ISDN protocol variant to be used.

Allowed Values:

- None = None (Standard)
- Alcatel = Support Alcatel Extensions.
- Ericsson = Support Ericsson Extensions.

Default Value = None

INI File Parameter Name = t1IsdnProtocolVariant

3.7.3.3 Line Encoding

Description: Specifies the type of T1 line coding that will be used.

Allowed Values:

- AMI = Alternate Mark Inversion line coding is used.
- B8ZS = Binary Eight Zero Substitution line coding is used.

Default Value = B8ZS

INI File Parameter Name = t1Encoding

3.7.3.4 Framing

Description: Specifies the type of T1 framing that will be used by the line.

Allowed Values:

- SF = D4 Superframe format (12 consecutive T1 frames) is used.
- ESF = Extended Superframe format (24 consecutive T1 frames) is used.

Default Value = ESF

INI File Parameter Name = t1Framing

3.7.3.5 Selects Transmit Pulse Waveform

Description: Selects the method to be used for generating the Transmit Waveform Shape.

Allowed Values:

- Short Haul 110 ft = The DMG2000 is 0 - 110 feet from T1 line source.
- Short Haul 220 ft = The DMG2000 is 110 - 220 feet from T1 line source.
- Short Haul 330 ft = The DMG2000 is 220 - 330 feet from T1 line source.
- Short Haul 440 ft = The DMG2000 is 330 - 440 feet from T1 line source.
- Short Haul 550 ft = The DMG2000 is 440 - 550 feet from T1 line source.
- Short Haul 660 ft = The DMG2000 is 550 - 660 feet from T1 line source.

Default Value = Short Haul 110 ft

INI File Parameter Name = t1TxWave

3.7.3.6 Outbound TDM Calling Party Source

Description: The source of the calling party that will be provided to the TDM interface for outbound calls and MWI requests.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for T1 QSIG, this parameter must be set to one of the following options in conjunction with the Static TDM Calling Party parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values:

- None = No calling party is sent.
- VoIP = Calling party is derived from the VoIP call/mwi-request.
- Static = A statically configured calling party is used.

- VoIP_Preferred = Calling party is derived from the VoIP call/mwi-request. If not available from VoIP, then the statically configured calling party is used.

Default Value = Static

INI File Parameter Name = gwTdmOutboundCallingPartySrc

3.7.3.7 Static TDM Calling Party

Description: Specifies the calling party that will be provided to the TDM interface if Outbound TDM Calling Party Source is set to Static or VoIP_Preferred.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for T1 QSIG, this parameter must be set to one of the following options in conjunction with the Outbound TDM Calling Party Source parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values: Any valid dialable number

Default Value = 1234

INI File Parameter Name = gwTdmOutboundCallingParty

3.7.4 E1 ISDN Protocol Group (ISDN Signaling Mode)

The parameters in the E1 ISDN Protocol group include the following:

- ISDN Protocol
- ISDN Protocol Variant
- Line Encoding
- Framing
- Selects Transmit Pulse Waveform
- Contiguous B-Channel
- Outbound TDM Calling Party Source
- Static TDM Calling Party

3.7.4.1 ISDN Protocol

Description: If ISDN is selected as the Signaling Mode, specifies the ISDN protocol to be used.

Allowed Values:

- QSIG = QSIG ISDN protocol is used.
- ETSI = EuroISDN (ETSI) protocol is used.

Default Value = QSIG

INI File Parameter Name = e1IsdnProtocol

Note: Unit requires a restart if this parameter value is changed.

3.7.4.2 ISDN Protocol Variant

Description: Specifies the type of E1 ISDN protocol variant to be used.

Allowed Values:

- None = None (Standard)
- Alcatel = Support Alcatel Extensions.
- Ericsson = Support Ericsson Extensions.

Default Value = None

INI File Parameter Name = e1IsdnProtocolVariant

3.7.4.3 Line Coding

Description: Specifies the type of T1 line coding that will be used.

Allowed Values:

- AMI = Alternate Mark Inversion line coding is used.
- HDB3 = High Density Bipolar Three line coding is used.

Default Value = HDB3

INI File Parameter Name = e1Encoding

Note: Unit requires a restart if this parameter value is changed.

3.7.4.4 Framing

Description: Specifies the type of E1 framing that will be used by the line.

Allowed Values:

- CRC_MF = Multiframe format with CRC is used.
- FR = Basic frame format is used.
- MF = Multiframe format is used.

Default Value =CRC_FMF

INI File Parameter Name = e1Framing

Note: Unit requires a restart if this parameter value is changed.

3.7.4.5 Selects Transmit Pulse Waveform

Description: Selects the method to be used for generating the Transmit Waveform Shape.

Allowed Values:

- 75_Ohm = 75 ohm (unbalanced) impedance is used.
- 120_Ohm = 120 ohm (balanced) impedance is used.

Default Value = 75 Ohm Impedance

INI File Parameter Name = e1TxWave

Note: Unit requires a restart if this parameter value is changed.

3.7.4.6 Contiguous B-Channel

Description: Specifies B-Channel Selection Mode Used By Switch. Only applicable to E1 ISDN protocols.

Allowed Values:

- Yes = Contiguous B-Channel Selection (logical mapping).
- No = Skip B-Channel 16 (physical mapping)

Default Value = No

INI File Parameter Name = e1ContiguousBchan

3.7.4.7 Outbound TDM Calling Party Source

Description: The source of the calling party that will be provided to the TDM interface for outbound calls and MWI requests.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for E1 QSIG, this parameter must be set to one of the following options in conjunction with the Static TDM Calling Party parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values:

- None = No calling party is sent.
- VoIP = Calling party is derived from the VoIP call/mwi-request.
- Static = A statically configured calling party is used.
- VoIP_Preferred = Calling party is derived from the VoIP call/mwi-request. If not available from VoIP, then the statically configured calling party is used.

Default Value = Static

INI File Parameter Name = gwTdmOutboundCallingPartySrc

3.7.4.8 Static TDM Calling Party

Description: Specifies the calling party that will be provided to the TDM interface if Outbound TDM Calling Party Source is set to Static or VoIP_Preferred.

Note: When a DMG2000 is connected to a Mitel PBX and is configured for E1 QSIG, this parameter must be set to one of the following options in conjunction with the Outbound TDM Calling Party Source parameter:

- Outbound TDM Calling Party Source set to **Static** and Static TDM Calling Party set to a **number** (e.g. 1234).
- Outbound TDM Calling Party Source set to **VoIP** and the VoIP endpoint must provide a calling party number in the VoIP call/MWI call request.
- Outbound TDM Calling Party Source set to **VoIP_Preferred** and Static TDM Calling Party set to a **number** (e.g. 1234).

Allowed Values: Any valid dialable number

Default Value = 1234

INI File Parameter Name = gwTdmOutboundCallingParty

3.7.5 CPID Parsing Configuration

Description: Modifying this data affects either the in-band Type I (on-hook) or Type II (off-hook) CPID parsing of the DMG2000, depending on how the unit is configured. The text box on this page can be easily modified by either pasting parsing rules into it from a text file or directly modifying the contents of the control. See Chapter 6, "Media Gateway Parsers" for further information about the creation and use of parsing rules.

3.8 H.323 Parameters

The H.323 parameters include:

- Gatekeeper Mode
- Gateway Name

- Gateway Prefix
- Gatekeeper IP Address
- Gatekeeper Multicast IP Address
- Proxy IP Address
- Preferred Gatekeeper ID
- Fast Start
- H.245 Signaling User Input
- Master OLC Transmit Delay

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.1 Gatekeeper Mode

Description: Specifies the H.323 gatekeeper mode.

Allowed Values:

- None = No gatekeeper is used.
- Manual = Gatekeeper IP Address parameter is used to identify the IP address of the gatekeeper.
- Auto = Media Gateway searches for the gatekeeper itself.
- Proxy = No gatekeeper is used. Instead, the Media Gateway sends all requests to a Proxy Server (whose IP address is configured via the Proxy IP Address parameter).

Default Value = None

INI File Parameter Name = h323GKMode

Note: Unit requires a restart if this parameter value is changed.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.2 Gateway Name

Description: Specifies the H.323 Gateway alias name to be used by the Media Gateway.

Allowed Values: Any valid string (1 to 254 characters)

Default Value = PBX_GW

INI File Parameter Name = h323GatewayAlias

Note: Unit requires a restart if this parameter value is changed.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.3 Gateway Prefix

Description: Specifies the E.164 prefix number to be used by IP terminal devices to access this Media Gateway. Used only when an H.323 gatekeeper is in the system.

Allowed Values: Any valid E.164 prefix number (0 to 9 digits)

Default Value = 9

INI File Parameter Name = h323GatewayPrefix

Note: Unit requires a restart if this parameter value is changed.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.4 Gatekeeper IP Address

Description: Specifies the H.323 gatekeeper IP address. Used only if Gatekeeper Mode is Manual.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 10.10.11.254

INI File Parameter Name = h323GKAddr

Note: Unit requires a restart if this parameter value is changed.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.5 Gatekeeper Multicast IP Address

Description: Specifies the IP address used to broadcast a request for gatekeeper auto-discovery. Used only if Gatekeeper Mode is Auto.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = 255.255.255.255

INI File Parameter Name = h323GKMcastAddr

Note: Unit requires a restart if this parameter value is changed.

3.8.6 Proxy IP Address

Description: Specifies the IP address of the Proxy Server through which the Media Gateway will send all requests if the Gatekeeper Mode parameter is set to Proxy.

Allowed Values: Any valid IP address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = h323ProxyAddr

3.8.7 Preferred Gatekeeper ID

Description: Specifies the preferred gatekeeper ID when multiple gatekeepers respond to a gatekeeper auto-discovery query. Used only when the GateKeeper Mode is Auto.

Allowed Values: Any valid string (0 to 63 characters)

Default Value = (no default value)

INI File Parameter Name = h323GKPreferred

Note: Unit requires a restart if this parameter value is changed.

3.8.8 Fast Start

Description: Specifies if the Fast Start method will be attempted during H.323 call setup.

Allowed Values:

- Yes = Fast Start method is enabled.
- No = Fast Start method is not enabled.

Default Value = Yes

INI File Parameter Name = h323FastStart

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.8.9 H.245 Signaling User Input

Description: Specifies the format of the User-to-User Input messages that are transported between the Media Gateway and the VoIP endpoints.

Allowed Values:

- Yes = User-to-User Input messages are sent between the Media Gateway and the VoIP endpoints using the Signaling format.
- No = User-to-User Input messages are sent between the PBX-I Media Gateway and the VoIP endpoints using the Alpha/Numeric format.

Default Value = Yes

INI File Parameter Name = h323SignalingUUI

Note: This parameter is only valid when the Signaling Digit Relay parameter is set to On. If the Signaling Digit Relay parameter is set to Off, this parameter is ignored.

3.8.10 Master OLC Transmit Delay

Description: Specifies the delay in milliseconds before H.323 master transmits an Open-Logical-Channel.

Allowed Values: 0 through 30000

Default Value = 2000

INI File Parameter Name = h323OLCXmitDelayMs

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

3.9 Session Initiation Protocol (SIP) Parameters

The SIP parameters include the following groups:

- User-Agent Group
- Server Group
- TCP/UDP Group
- TLS Group
- Proxy Group
- Timing Group

3.9.1 User-Agent Group

The User-Agent group includes the following parameters:

- Host and Domain Name
- Transport Type
- Call as Domain Name?
- SIPS URI Scheme Enabled
- Invite Expiration

3.9.1.1 Host and Domain Name

Description: The host and domain name of the Media Gateway. This name is used when the unit registers with the SIP Registration Server.

Allowed Values: Domain Name string

Default Value = pbxgw.default.com

INI File Parameter Name = sipServerDomain

Note: Unit requires a restart if this parameter value is changed.

3.9.1.2 Transport Type

Description: Defines the preferred transport protocol of call signaling packets.

Allowed Values:

- TCP = Transmission Control Protocol is used as the transport protocol.
- UDP = User Datagram Protocol is used as the transport protocol.

Default Value = UDP

INI File Parameter Name = sipTransportType

3.9.1.3 Call as Domain Name?

Description: Defines the host name used in the From header of generated INVITE requests.

Allowed Values:

- Yes = The Media Gateway's domain name is used as the Host Name.
- No = The Media Gateway's IP address is used as the Host Name.

Default Value = No

INI File Parameter Name = sipCallAsDomainName

3.9.1.4 SIPS URI Scheme Enabled

Description: Defines whether the SIPS URI (Uniform Resource Identifier) or SIP URI scheme will be used for generating SIP messages.

Allowed Values:

- Yes = All Request, To, From, and Contact URIs generated by the gateway will use the SIPS URI scheme
- No = All Request, To, From, and Contact URIs generated by the gateway will use the SIP URI scheme.

Default Value = No

INI File Parameter Name = sipSipsUriEnabled

3.9.1.5 Invite Expiration

Description: Specifies the amount of time in seconds that an INVITE request sent by the Media Gateway is valid and can be accepted by the SIP endpoint. After the defined time in seconds, the INVITE request expires and is no longer valid.

Allowed Values: 1 - 60000 seconds

Default Value = 120 seconds

INI File Parameter Name = sipExpInvSec

3.9.2 Server Group

The Server group includes the following parameters:

- DNS Server Address
- DNS Translation of Phone Numbers
- Registration Server Address
- Registration Server Port
- Registration Expiration (sec)

3.9.2.1 DNS Server Address

Description: Specifies the IP address of the Domain Name Server (DNS) that the Media Gateway will use to resolve IP address information.

Notes:
1. If the DNS Server IP Address parameter is configured but the Primary Proxy Server Address parameter is not, the Media Gateway will use the DNS Server to resolve IP address information.
2. If both the DNS Server IP Address parameter and the Primary Proxy Server Address parameter are configured, the Media Gateway will use the Proxy Server for all requests.
3. If the DNS Server IP Address parameter is configured and the Primary Proxy Server Address parameter is configured with an alias, then the Media Gateway will first use the DNS Server to resolve the alias of the Proxy Server to an IP address and then use the Proxy Server for all subsequent requests.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = sipDnsServerAddr

3.9.2.2 DNS Translation of Phone Numbers

Description: If 'Yes', the Media Gateway will use DNS to translate tel URIs and URIs with the 'user=phone' parameter. If 'No', the Media Gateway will not use DNS to translate these types of URIs.

Allowed Values:

- Yes
- No

Default Value = No

INI File Parameter Name = sipEnumDnsEnabled

3.9.2.3 Registration Server Address

Description: IP Address of the SIP Registration Server that the Media Gateway should register with. If blank, the Media Gateway will not register with a Registration Server.

Allowed Values:

- Any valid IP address in dotted decimal notation.
- Blank

Default Value = (no default value)

INI File Parameter Name = sipRegAddr

Note: Unit requires a restart if this parameter value is changed.

3.9.2.4 Registration Server Port

Description: IP Port of the SIP Registration Server. If a SIP Registration Server IP Address was entered for the Registration Server Address parameter, then the Registration Server Port parameter must be set to a valid port number.

Allowed Values: 1024-65000

Default Value = 5060

INI File Parameter Name = sipRegPort

3.9.2.5 Registration Expiration (sec)

Description: Specifies the amount of time in seconds that the registration with the SIP Proxy Server is valid.

Allowed Values: 1 - 60000 seconds

Default Value = 3600 seconds

INI File Parameter Name = sipExpRegSec

3.9.3 TCP/UDP Group

The TCP/UDP group includes the following parameters:

- UDP/TCP Transport Enabled
- TCP/UDP Server Port
- TCP Inactivity Timer

3.9.3.1 UDP/TCP Transport Enabled

Description: Enables or disables the UDP/TCP transports.

Allowed Values:

- Yes = UDP and TCP transports are enabled.
- No = UDP and TCP transports are disabled.

Default Value = Yes

INI File Parameter Name = sipUdpTcpEnabled

Note: Unit requires a restart if this parameter value is changed.

3.9.3.2 TCP/UDP Server Port

Description: The TCP/UDP Port of the Media Gateway on which SIP messages are sent/received.

Allowed Values: 1024 - 65000

Default Value = 5060

INI File Parameter Name = sipServerPort

Note: Unit requires a restart if this parameter value is changed.

3.9.3.3 TCP Inactivity Timer

Description: Number of seconds after which an idle Transmission Control Protocol (TCP) connection will be closed.

Allowed Values: 10 - 60000 seconds

Default Value = 30 seconds

INI File Parameter Name = sipTcpInactivitySec

3.9.4 TLS Group

The TLS group includes the following parameters:

- TLS Transport Enabled
- TLS Server Port
- SSL TLS Protocol
- Mutual Authentication Required
- TLS Inactivity Timer
- Verify TLS Peer Certificate Date

- Verify TLS Peer Certificate Trust

3.9.4.1 TLS Transport Enabled

Description: Determines whether the TLS transport is enabled or disabled.

Allowed Values:

- Yes = TLS transport is enabled.
- No = TLS transport is disabled.

Default Value = No

INI File Parameter Name = sipTlsEnabled

Note: Unit requires a restart if this parameter value is changed.

3.9.4.2 TLS Server Port

Description: The Transport Layer Security (TLS) Port of the Media Gateway on which SIP TLS messages are sent/received.

Allowed Values: 1024 - 65000

Default Value = 5061

INI File Parameter Name = sipTlsServerPort

Note: Unit requires a restart if this parameter value is changed.

3.9.4.3 SSL TLS Protocol

Description: Specifies the default SSL record type to be used on TLS connections.

Allowed Values:

- SSLv3_TLSv1 = Connection will understand both the SSLv3 and TLSv1 protocols.
- SSLv3_Only = Connection will only understand the SSLv3 protocol.
- TLSv1_Only = Connection will only understand the TLSv1 protocol.

Default Value = SSLv3_TLSv1

INI File Parameter Name = secSipTlsProtocol

Note: Unit requires a restart if this parameter value is changed.

3.9.4.4 Mutual Authentication Required

Description: If Yes is selected, the SIP client sends a certificate to the peer server and the SIP server will send a certificate request to the client. The certificate received from the client is validated. If the client does not provide a valid certificate, the server will close the session.

If No is selected, the SIP server does not send a client certificate request to the client and the client does not send a certificate to the server. A certificate received from the client is ignored by the server.

Allowed Values:

- Yes = SIP client sends certificate to peer server and SIP server sends certificate request to client.
- No = SIP server does not send client a certificate request and client does not send a certificate to the server.

Default Value = Yes

INI File Parameter Name = sipTlsMutualAuthentication

Note: Unit requires a restart if this parameter value is changed.

3.9.4.5 TLS Inactivity Timer

Description: Number of seconds after which an idle Transport Layer Security (TLS) connection will be closed.

Allowed Values: 10 - 60000 seconds

Default Value = 30 seconds

INI File Parameter Name = sipTlsInactivitySec

3.9.4.6 Verify TLS Peer Certificate Date

Description: Determines whether the TLS peer's certificate date is validated. This requires that the Media Gateway has already received the network time via SNTP.

Allowed Values:

- Yes = Validate certificate date
- No = Do not validate certificate date

Default Value = Yes

INI File Parameter Name = sipTlsCertVerifyDate

3.9.4.7 Verify TLS Peer Certificate Trust

Description: Determines whether the TLS peer's certificate trust relationship is validated against the list of trusted CA certificates.

Allowed Values:

- Yes = Validate
- No = Do not validate

Default Value = Yes

INI File Parameter Name = sipTlsCertVerifyTrust

3.9.5 Proxy Group

The Proxy group includes the following parameters:

- Primary Proxy Server Address
- Primary Proxy Server Port
- Backup Proxy Server Address
- Backup Proxy Server Port
- Proxy Query Interval (sec)

3.9.5.1 Primary Proxy Server Address

Description: The IP Address of the SIP Proxy Server through which the Media Gateway may send/receive requests. If blank, the Media Gateway will not use a Proxy Server.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = sipProxyServerAddr

3.9.5.2 Primary Proxy Server Port

Description: The IP Port of the SIP Proxy Server. If an IP Address was entered for the Primary Proxy Server Address parameter, then the Primary Proxy Server Port parameter must be set to a valid port number.

Allowed Values: 1024-65000

Default Value = 5060

INI File Parameter Name = sipProxyServerPort

3.9.5.3 Backup Proxy Server Address

Description: The IP Address of the SIP Backup Proxy Server through which the Media Gateway may send/receive requests. If blank, the Media Gateway will not use a Backup Proxy Server.

Allowed Values: Any valid IP Address in dotted decimal notation.

Default Value = (no default value)

INI File Parameter Name = sipProxyServerAddr2

3.9.5.4 Backup Proxy Server Port

Description: The IP Port of the SIP Backup Proxy Server. If an IP Address was entered for the Backup Proxy Server Address parameter, then the Backup Proxy Server Port parameter must be set to a valid port number.

Allowed Values: 1024-65000

Default Value = 5060

INI File Parameter Name = sipProxyServerPort2

3.9.5.5 Proxy Query Interval (sec)

Description: Interval in seconds at which the Primary Proxy Server is queried. If the Primary Proxy Server does not respond to the query, the Media Gateway will switch to the Backup Proxy Server. Once the Primary Proxy Server responds to the query, the Media Gateway will switch back to the Primary Proxy Server.

Note: This parameter is only valid if the Backup Proxy Server Address parameter is configured. The Primary Proxy Server must respond to SIP OPTIONS requests in order for the Proxy Query to succeed.

Allowed Values: 10-3600 seconds

Default Value = 30 seconds

INI File Parameter Name = sipProxyQueryIntervalSec

3.9.6 Timing Group

The Timing group includes the following parameters:

- T1 Time
- T2 Time
- T4 Time

3.9.6.1 T1 Time

Description: The T1 Time specifies the SIP request retransmit timeout in milliseconds. This timer is started when a SIP request is generated. If no response to the request is received in T1 Time milliseconds, the request is retransmitted and the timeout is doubled to 2*T1 Time milliseconds. If again no response is received before the new timeout, the message is again retransmitted and the timeout is again doubled - this time to 4*T1 Time milliseconds.

Allowed Values: 200 - 60000 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = sipT1TimeMs

3.9.6.2 T2 Time

Description: The T2 Time specifies the maximum retransmit time of SIP request messages (except for INVITE) in milliseconds. The retransmitting of the request and the doubling of the timeout continues until the timeout reaches T2 Time milliseconds. Once the timeout reaches T2 Time, the request is retransmitted at T2 Time intervals until a response is received or until the message has been retransmitted "NumRetriesRequest" times - at which point the request expires.

Allowed Values: 200 - 60000 milliseconds

Default Value = 4000 milliseconds

INI File Parameter Name = sipT2TimeMs

3.9.6.3 T4 Time

Description: The T4 Time specifies the time in milliseconds that the network will take to clear messages between client and server transactions.

Allowed Values: 1000 - 60000 milliseconds

Default Value = 5000 milliseconds

INI File Parameter Name = sipT4TimeMs

3.10 Non-Menu (Hidden) Parameters

The following parameters are not included in the Configure Web pages and cannot be accessed from the Configuration menu selections. Modifying any of these parameters significantly changes the operation of the Media Gateway. The non-menu configuration parameters include the following groups:

- DSP Advanced Group
- CPID Parsing Configuration

3.10.1 DSP Advanced Group

The DSP Advanced parameters are not included in the Configuration menu selections. These parameters are accessed instead by typing **dsp.htm** following the Media Gateway IP address in the Web browser address box. For example:

```
http://10.12.13.74/dsp.htm
```

Caution: Modifying any of these parameters affects the digital signal processors and significantly changes the operation of the Media Gateway.

Caution: The DSP Advanced configuration parameters should only be modified by advanced users of the Media Gateway. You are strongly advised to backup your current configuration before proceeding. See Section 2.6.1, "Exporting Configuration Information", on page 39 for additional information about backing up your configuration.

DSP Advanced Settings

- TDM to IP Gain Adjustment (DMG1000 Only)
- IP to TDM Gain Adjustment (DMG1000 Only)
- Line Echo Cancellation (DMG1000 Only)
- Line Echo Cancellation NLP (DMG1000 Only)
- Line Echo Cancellation NLP Aggressiveness
- Voice Activity Noise Floor
- Voice Activity Measurement Period (DMG2000 Only)
- Voice Activity Signal to Noise Ratio (DMG2000 Only)
- Call Progress Filter Threshold
- Call Progress Filter Debounce
- Call Progress Filter Percent (DMG1000 Only)
- Call Progress Filter Low Cutoff (DMG1000 Only)
- Call Progress Filter High Cutoff (DMG1000 Only)
- Call Progress Filter SNR in dB (DMG2000 Only)
- Call Progress Filter Two Tones Max Twist in dB (DMG2000 Only)
- Jitter-Buffer Minimum Delay (DMG2000 Only)
- Jitter-Buffer Maximum Delay (DMG2000 Only)
- Jitter-Buffer Initial Delay (DMG2000 Only)
- Jitter-Buffer Adaptation Period (DMG2000 Only)
- Jitter-Buffer Deletion Threshold (DMG2000 Only)
- Jitter-Buffer Frame Deletion Mode (DMG2000 Only)
- IP to PCM AGC Enable (DMG2000 Only)
- IP to PCM AGC Slew Rate (DMG2000 Only)
- IP to PCM AGC Target Level (DMG2000 Only)

- IP to PCM AGC Max Gain (DMG2000 Only)
- IP to PCM AGC Min Gain (DMG2000 Only)
- TDM to IP AGC Enable (DMG2000 Only)
- TDM to IP AGC Slew Rate (DMG2000 Only)
- TDM to IP AGC Target Level (DMG2000 Only)
- TDM to IP AGC Max Gain (DMG2000 Only)
- TDM to IP AGC Min Gain (DMG2000 Only)

T.38 Advanced Settings

- Allow T.38 ECM Faxes (DMG2000 Only)
- Transmit Small T4 ECM T.38 Packets (DMG2000 Only)
- Enable T.38 Spoofing (DMG2000 Only)
- TSI Removal (DMG2000 Only)
- CSI Removal (DMG2000 Only)
- NSF Removal (DMG2000 Only)
- DIS Removal (DMG2000 Only)
- T.38 Packet Loss Concealment Method (DMG2000 Only)
- T.38 UDPTL Redundancy Count for Fax Page Data (DMG2000 Only)
- T.38 UDPTL Redundancy Count for T.30 Messages (DMG2000 Only)
- Fax Transmit Level (dBm) (DMG2000 Only)
- Maximum UDPTL Packet Size (DMG2000 Only)

3.10.2 DSP Advanced Settings

3.10.2.1 TDM to IP Gain Adjustment (DMG1000 Only)

Description: Adjusts the gain of the audio signal in the Telephony-to-IP direction.

Allowed Values: -14 dB to +14 dB

Default Value = 0 dB

INI File Parameter Name = dspPbxToIPGain

3.10.2.2 IP to TDM Gain Adjustment (DMG1000 Only)

Description: Adjusts the gain of the audio signal in the IP-to-Telephony direction.

Allowed Values: -14 dB to +14 dB

Default Value = 0 dB

INI File Parameter Name = dspIPtoPbxGain

3.10.2.3 Line Echo Cancellation (DMG1000 Only)

Description: Enables or disables the echo canceller on the PBX (TDM) side.

Allowed Values:

- On = Enables the echo canceller.
- Off = Disables the echo canceller.

Default Value = On

INI File Parameter Name = dspEc

3.10.2.4 Line Echo Cancellation NLP (DMG1000 Only)

Description: Enables or disables the echo canceller non-linear processor (NLP) on the PBX (TDM) side.

Allowed Values:

- On = Enables the echo canceller NLP.
- Off = Disables the echo canceller NLP.

Default Value = On

INI File Parameter Name = dspEcNlp

3.10.2.5 Line Echo Cancellation NLP Aggressiveness

Description: This parameter controls the aggressiveness of the echo canceller NLP.

Allowed Values for DMG1000: -32768 to +32767

Default Value = 0

Allowed Values for T1/E1 Media Gateway:

- Normal = G.168 compliant
- Increased = Maximum non-linear echo protection
- Reduced = Provides the least double talk distortion

Default Value = Normal

INI File Parameter Name = dspEcNlpAgg

Note: Unit requires a restart if this parameter value is changed.

3.10.2.6 Voice Activity Noise Floor

Description: Defines the noise floor for the voice activity detector (VAD). Signal levels below the value selected will be treated as silence by the VAD.

Allowed Values: -80 dB to -10 dB

Default Value = -40 dB

INI File Parameter Name = vadNoiseFloor

Note: Unit requires a restart if this parameter value is changed.

3.10.2.7 Voice Activity Measurement Period (DMG2000 Only)

Description: Specifies the Measurement Period for the Voice Activity Detector. The Voice Activity Detector will wait at least this long to determine whether the signal on the line is voice.

Allowed Values: 20 to 200

Default Value = 30

INI File Parameter Name = vadMeasurementPeriod

3.10.2.8 Voice Activity Signal to Noise Ratio (DMG2000 Only)

Description: Specifies the Signal to Noise Ration (SNR) for the Voice Activity Detector. Tonal signals with an SNR greater than this value will be treated as tones and ignored by the Voice Activity Detector.

Allowed Values: 10 to 20

Default Value = 18

INI File Parameter Name = vadMinSNR

3.10.2.9 Call Progress Filter Threshold

Description: Defines the lower threshold for the Call Progress Tone Detector. Call progress signals below this value will be ignored.

Allowed Values: -38 dB to -20 dB

Default Value = -30 dB

INI File Parameter Name = cpFilterThreshold

Note: Unit requires a restart if this parameter value is changed.

3.10.2.10 Call Progress Filter Debounce

Description: Specifies the debounce time in milliseconds for the Call Progress Tone Detector. This value defines the minimum time that a call progress signal must be present before the detector is triggered.

Allowed Values: 100 to 32768 milliseconds

Default Value = 100 milliseconds

INI File Parameter Name = cpFilterDebounce

Note: Unit requires a restart if this parameter value is changed.

3.10.2.11 Call Progress Filter Percent (DMG1000 Only)

Description: Specifies the percent ratio between tone power and total power for the Call Progress Tone Detector. Call progress tone signals whose power is a smaller percentage of total power than this value will be ignored.

Allowed Values: 25 to 87%

Default Value = 45%

INI File Parameter Name = cpFilterPercent

Note: Unit requires a restart if this parameter value is changed.

3.10.2.12 Call Progress Filter Low Cutoff (DMG1000 Only)

Description: Specifies the low cut-off frequency for the Call Progress Tone Detector. Call progress signals below this frequency will be ignored.

Allowed Values: 250 Hz to 1000 Hz

Default Value = 300 Hz

INI File Parameter Name = cpFilterLowCuttoff

Note: Unit requires a restart if this parameter value is changed.

3.10.2.13 Call Progress Filter High Cutoff (DMG1000 Only)

Description: Specifies the high cut-off frequency for the Call Progress Tone Detector. Call progress signals above this frequency will be ignored.

Allowed Values: 500 Hz to 2000 Hz

Default Value = 650 Hz

INI File Parameter Name = cpFilterHighCutoff

Note: Unit requires a restart if this parameter value is changed.

3.10.2.14 Call Progress Filter SNR in dB (DMG2000 Only)

Description: Specifies the Signal to Noise Ratio (SNR) of the Call Progress Tone Detector. Call progress signals whose SNR is lower than specified will be ignored.

Allowed Values: 0 dB to 40 dB

Default Value = 20 dB

INI File Parameter Name = cpFilterSnr

Note: Unit requires a restart if this parameter value is changed.

3.10.2.15 Call Progress Filter Two Tones Max Twist in dB (DMG2000 Only)

Description: Specifies the maximum twist for two tone Call Progress Tones. Call progress signals that consist of two tones whose power ratio of the tones is greater than the specified twist will be ignored.

Allowed Values: 0 dB to 12 dB

Default Value = 3 dB

INI File Parameter Name = cpFilterMaxTwist

Note: Unit requires a restart if this parameter value is changed.

3.10.2.16 Jitter-Buffer Minimum Delay (DMG2000 Only)

Description: Specifies the minimum jitter-buffer delay. The smaller the delay, the smaller the jitter-buffer. It is recommended that this value be at least the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 20 milliseconds

INI File Parameter Name = dspJbDelayMinMs

3.10.2.17 Jitter-Buffer Maximum Delay (DMG2000 Only)

Description: Specifies the maximum jitter-buffer delay. A larger delay provides protection against larger network jitter, but increases audio delay. It is recommended that this value be set to at least four times the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 200 milliseconds

INI File Parameter Name = dspJbDelayMaxMs

3.10.2.18 Jitter-Buffer Initial Delay (DMG2000 Only)

Description: Specifies the starting jitter-buffer delay. It is recommended that this value be set to at least two times the duration of one RTP packet interval. If the minimum, maximum, and initial jitter-buffer delays are set to the same value, the jitter-buffer will operate in the non-adaptive mode.

Allowed Values: 0 to 200 milliseconds

Default Value = 20 milliseconds

INI File Parameter Name = dspJbDelayInitMs

3.10.2.19 Jitter-Buffer Adaptation Period (DMG2000 Only)

Description: Controls the speed at which the jitter-buffer adapts downward when network conditions allow. The larger the value, the slower the jitter-buffer adapts downward when jitter decreases.

Allowed Values: 1000 to 65535 milliseconds

Default Value = 10000 milliseconds

INI File Parameter Name = dspJbAdaptationPeriodMs

3.10.2.20 Jitter-Buffer Deletion Threshold (DMG2000 Only)

Description: When the jitter-buffer grows past this value, frames exceeding the threshold are deleted immediately.

Allowed Values: 0 to 500 milliseconds

Default Value = 500 milliseconds

INI File Parameter Name = dspJbDeletionThresholdMs

3.10.2.21 Jitter-Buffer Frame Deletion Mode (DMG2000 Only)

Description: Determines how frames are deleted when the jitter-buffer adapts downward.

Allowed Values:

- Soft = More emphasis on audio quality, but maximum delay may be exceeded.
- Hard = Hard cap on the maximum delay, which may negatively impact audio quality.

Default Value = Soft

INI File Parameter Name = dspJbDeletionMode

3.10.2.22 IP to PCM AGC Enable (DMG2000 Only)

Description: Enables automatic gain control (AGC) in the IP to PCM direction.

Allowed Values:

- On = AGC is enabled.
- Off = AGC is disabled.

Default Value = On

INI File Parameter Name = dspIpToPbxAgcEnable

3.10.2.23 IP to PCM AGC Slew Rate (DMG2000 Only)

Description: Determines the speed at which the AGC adapts in the IP to PCM direction.

Allowed Values:

- Fast = More emphasis on AGC gain adjustments
- Medium = Mid point
- Slow = More emphasis on Audio Quality

Default Value = Medium

INI File Parameter Name = dspIpToPbxAgcSlewRate

3.10.2.24 IP to PCM AGC Target Level (DMG2000 Only)

Description: Defines the IP to PCM AGC target level for a range of -50 to 0 dBm.

Allowed Values: -50 to 0 dBm

Default Value = -14 dBm

INI File Parameter Name = dspIpToPbxAgcLevel

3.10.2.25 IP to PCM AGC Max Gain (DMG2000 Only)

Description: Defines the IP to PCM AGC maximum gain within the range of 0 to 15 dB.

Allowed Values: 0 to 15 dB

Default Value = 12 dB

INI File Parameter Name = dspIpToPbxAgcMaxGain

3.10.2.26 IP to PCM AGC Min Gain (DMG2000 Only)

Description: Defines the IP to PCM AGC minimum gain within the range of 0 to -15 dB.

Allowed Values: 0 to -15 dB

Default Value = -15 dB

INI File Parameter Name = dspIpToPbxAgcMinGain

3.10.2.27 TDM to IP AGC Enable (DMG2000 Only)

Description: Enables automatic gain control (AGC) in the PCM to IP direction.

Allowed Values:

- On = AGC is enabled.
- Off = AGC is disabled.

Default Value = On

INI File Parameter Name = dspPbxToIpAgcEnable

3.10.2.28 TDM to IP AGC Slew Rate (DMG2000 Only)

Description: Determines the speed at which the AGC adapts in the PCM to IP direction.

Allowed Values:

- Fast = More emphasis on AGC gain adjustments
- Medium = Mid point
- Slow = More emphasis on Audio Quality

Default Value = Medium

INI File Parameter Name = dspPbxToIpAgcSlewRate

3.10.2.29 TDM to IP AGC Target Level (DMG2000 Only)

Description: Defines the PCM to IP AGC target level for a range of -50 to 0 dBm.

Allowed Values: -50 to 0 dBm

Default Value = -14 dBm

INI File Parameter Name = dspPbxToIpAgcLevel

3.10.2.30 TDM to IP AGC Max Gain (DMG2000 Only)

Description: Defines the PCM to IP AGC maximum gain within the range of 0 to 15 dB.

Allowed Values: 0 to 15 dB

Default Value = 12 dB

INI File Parameter Name = dspPbxToIpAgcMaxGain

3.10.2.31 TDM to IP AGC Min Gain (DMG2000 Only)

Description: Defines the PCM to IP AGC minimum gain within the range of 0 to -15 dB.

Allowed Values: 0 to -15 dB

Default Value = -15 dB

INI File Parameter Name = dspPbxToIpAgcMinGain

3.10.3 T.38 Advanced Settings

3.10.3.1 Allow T.38 ECM Faxes (DMG2000 Only)

Description: Enables/disables Error Correction Mode faxes in T.38.

Allowed Values:

- On = ECM allowed
- Off = ECM disabled

Default Value = On

INI File Parameter Name = dspT38EcmEnable

3.10.3.2 Transmit Small T4 ECM T.38 Packets (DMG2000 Only)

Description: Enables/disables small T4 ECM packet transmission in T.38.

Allowed Values:

- On = Enable small (<256 byte) T4 ECM packet transmission.
- Off = Wait for complete HDLC ECM frame from PCM before processing.

Default Value = On

INI File Parameter Name = dspT38EcmSmallPacketEnable

3.10.3.3 Enable T.38 Spoofing (DMG2000 Only)

Description: Enables/disables spoofing in T.38.

Allowed Values:

- On = Enable T.38 spoofing.
- Off = Disable T.38 spoofing.

Default Value = On

INI File Parameter Name = dspT38SpoofingEnable

3.10.3.4 TSI Removal (DMG2000 Only)

Description: Determines if the Transmitting Subscriber Identification (TSI) is removed.

Allowed Values:

- On - Remove TSI from TDM to reduce packet delay.
- Off - Include TSI in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38TsiRemovalEnable

3.10.3.5 CSI Removal (DMG2000 Only)

Description: Determines if the Called Subscriber Identification (CSI) is removed.

Allowed Values:

- On - Remove CSI from TDM to reduce packet delay.
- Off - Include CSI in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38CsiRemovalEnable

3.10.3.6 NSF Removal (DMG2000 Only)

Description: Determines if Non-Standard Facilities (NSF) is removed.

Allowed Values:

- On - Remove NSF from TDM to reduce packet delay.
- Off - Include NSF in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38NsfRemovalEnable

3.10.3.7 DIS Removal (DMG2000 Only)

Description: Determines if the Digital Identification signal (DIS) is removed.

Allowed Values:

- On - Remove DIS from TDM to reduce packet delay.
- Off - Include DIS in packet sent to packet network.

Default Value = Off

INI File Parameter Name = dspT38DisRemovalEnable

3.10.3.8 T.38 Packet Loss Concealment Method (DMG2000 Only)

Description: Selects the packet loss concealment method in T.38.

Allowed Values:

- None - No T.38 Packet Concealment.
- White_Line - Replace bad line with white line.
- Last_Good_Line - Replace bad line with last good line.

Default Value = White_Line

INI File Parameter Name = dspT38PacketLossConcealment

3.10.3.9 T.38 UDPTL Redundancy Count for Fax Page Data (DMG2000 Only)

Description: Specifies the number of redundant IFP frames transferred with each UDPTL packet.

Allowed Values: 0 to 3

Default Value = 2

INI File Parameter Name = dspUdptlDataRedundancy

3.10.3.10 T.38 UDPTL Redundancy Count for T.30 Messages (DMG2000 Only)

Description: Specifies the number of redundant IFP frames transferred with each UDPTL packet.

Allowed Values: 0 to 3

Default Value = 2

INI File Parameter Name = dspUdptlT30Redundancy

3.10.3.11 Fax Transmit Level (dBm) (DMG2000 Only)

Description: Set the FAX transmit level in T.38 FAX relay mode.

Allowed Values: -15 to 0

Default Value = -10

INI File Parameter Name = dspT38FaxTransmitLevel

3.10.3.12 Maximum UDPTL Packet Size (DMG2000 Only)

Description: Sets the maximum UDPTL packet size in bytes that the T.38 channel will transmit.

Note: This does not include the IP header or UDP header. It only includes the UDPTL header and UPDTL payload.

Allowed Values: 272 to 512

Default Value = 512

INI File Parameter Name = dspT38UdptlPacketSize

3.10.4 CPID Parsing Configuration

The CPID Parsing Configuration data is included for T1 CAS in the DMG2000 unit and in the Analog DMG1000 unit. For other models, this data can be accessed instead by typing cpid.htm following the Media Gateway IP address in the Web browser address box. For example:

http://10.12.13.74/cpid.htm

Modifying this data affects the display parsing of the Media Gateway.

Caution: The CPID Parsing Configuration data should only be modified by advanced users of the Media Gateway. You are strongly advised to backup your current configuration before proceeding. See Section 2.6.1, "Exporting Configuration Information", on page 39 for additional information about backing up your configuration.

The text box on this page can be easily modified by either pasting parsing rules into it from a text file or directly modifying the contents of the control. See Chapter 6, "Media Gateway Parsers" for further information on the creation and use of parsing rules.

INI File Parameter Name = ;-CPID RULES

3.11 Tone Parameters

The tone parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Media Gateway detects during call progress analysis.

These call progress tones can be viewed and edited manually from the Web interface if the characteristics are known, or the Media Gateway can be directed to analyze and learn the characteristics of specific call progress tones. For more information on viewing, editing, learning, and validating call progress tones from the Web interface, see Chapter 4, "Working with Call Progress Tones".

The call progress tone parameters include the following:

- ID
- Tone Event
- Tone Name
- Cadence Type
- Number of Cadence Cycles
- Tone Frequency
- Tone Frequency Deviation
- Tone Cadence Time
- Tone Cadence Time Deviation

In addition, you can edit the INI file directly if needed as discussed in the following topic:

- Editing the INI File Directly

3.11.1 ID

Description: Automatically assigned IDs for reference. These IDs cannot be edited.

3.11.2 Tone Event

Description: Specifies the tone event for this tone definition.

Allowed Values:

- None
- Dialtone
- Ringback
- Busy
- Congestion
- Disconnect
- Error
- SIT
- Pager
- Modem
- Fax
- FaxCNG

Default Value = None

INI File Parameter Name = cpToneEvent

3.11.3 Tone Name

Description: Specifies the text name associated with this tone definition.

Allowed Values: String with length between 0 - 31 characters

Default Value = Empty string

INI File Parameter Name = cpToneName

3.11.4 Cadence Type

Description: Specifies the cadence type for this tone definition.

Allowed Values:

- Continuous = an "on" tone with no "off".
- OnOff = an "on" cadence followed by an "off" cadence.
- OnOffQuick = detection is reported at the end of the "off" period following the first valid "on" period.
- Alternating = two tones that alternate with no "off" time in between.
- Sequential = one, two or three tones that come in sequence.

Default Value = Continuous

INI File Parameter Name = cpCadenceType

3.11.5 Number of Cadence Cycles

Description: Specifies the number of cadence cycles for this tone definition.

Allowed Values: 1-10

Default Value = 2

INI File Parameter Name = cpCadenceNumCycles

3.11.6 Tone Frequency

Description: Specifies the frequencies for this tone definition. A tone can be described by up to three (3) frequencies. For Alternating and Sequential Tones, the top frequency is the first frequency,

the middle frequency is the second frequency, and the bottom frequency is the third frequency. A value of 0 means to ignore the entry.

Allowed Values: 0-2000 Hz

Default Value = 0

INI File Parameter Name = cpToneFreq

3.11.7 Tone Frequency Deviation

Description: Specifies the deviation for the tone frequencies for this tone definition.

Allowed Values: 0-200 Hz

Default Value = 0

INI File Parameter Name = cpToneFreqDeviation

3.11.8 Tone Cadence Time

Description: Specifies the cadence times for this tone definition. A tone can be described by up to three (3) cadence times.

The interpretation of each time parameter is dependent on the selection of Cadence Type. For a given cadence type, Cadence Time specifies the time in milliseconds.

- Continuous - The first time value specifies the minimum time on that must elapse before the tone on event is generated.
- OnOff - The first two time values are used to specify the on time and off time respectively.
- OnOffQuick - The first time value is used to specify the minimum on time that causes detection and the second time value specifies the maximum off time that causes a cadence break.
- Alternating - The first two time values specify the on time for the corresponding frequency component.
- Sequential - For each of the tones in the sequence the corresponding time value specifies the on-time for the tone.

Allowed Values: 0-10000 ms

Default Value = 1500

INI File Parameter Name = cpToneTime

3.11.9 Tone Cadence Time Deviation

Description: Specifies the deviation for the cadence times for this tone definition.

Allowed Values: 0-1000 ms

Default Value = 700

INI File Parameter Name = cpToneTimeDeviation

3.11.10 Editing the INI File Directly

The call progress tone parameters can be edited directly in the INI file if desired. In older versions of the INI file, 16 tone definitions were listed regardless of the number of actual tone definitions in use. In these older INI files unused tone definitions would have the cpToneEvent parameter set to none. In newer versions of the INI file only the tone definitions in use are listed.

To delete a tone from the INI file, set the cpToneEvent to none, and that tone will be ignored when the INI file is read.

To add a tone to an INI file, either edit a tone definition whose cpToneEvent is set to none, or add a new tone definition. Then edit or add all the associated tone parameters.

3.12 Configuration File Parameters

The following parameters do not appear in the Web interface and can only be changed by editing the configuration file directly. These parameters are listed alphabetically.

- Auto-Answer Inbound TDM Calls (Type II CPID)
- Delay After Flash-Hook (msecs)
- Inform On PBX Calling Number Update (Phone Emulating Only)
- Inform On No PBX CPID (Phone Emulating Only)
- Inform On No PBX CPID Time (Phone Emulating Only)
- ISDN Overlap Receive Minimum Digits
- ISDN Overlap Receive Timeout (msecs)
- RFC 3960 Early Media Support
- Start Port for RTP
- Unauthenticated SRTP Enable
- UnEncrypted SRTCP Enable
- UnEncrypted SRTP Enable
- Wait for Dial Tone after Flash Hook

3.12.1 Auto-Answer Inbound TDM Calls (Type II CPID)

Description: Type II CPID requires that the inbound TDM call be answered by the gateway before the switch will send the inband CPID to the gateway. If this parameter is enabled, then the gateway will answer the call and gather the inband CPID from the switch prior to sending a call-request to

the VoIP endpoint. This allows the gateway to send the CPID information as part of the initial call-request to the VoIP endpoint. If this parameter is disabled, then the gateway will not answer the inbound TDM call until after the call-request is sent to the VoIP endpoint and the VoIP endpoint answers the call-request. In this case, the CPID information will not be part of the initial call-request to the VoIP endpoint.

Note: This parameter is valid only when the gateway is configured to receive Type II CPID (post-answer CPID).

If this parameter is disabled, the gateway will not answer the inbound TDM call until after the call-request is sent to the VoIP endpoint and the VoIP endpoint answers the call-request. In this case, the CPID information will not be part of the initial call-request to the VoIP endpoint.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Gateway will auto-answer the inbound TDM call.
- No = Gateway will not auto-answer the inbound TDM call.

Default Value = No

INI File Parameter Name = telAutoAnswer

3.12.2 Delay After Flash-Hook (msecs)

Description: Specifies the duration (in milliseconds) to delay after a Flash Hook if a dial tone confirmation is not expected.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 100 to 10000 msec

Default Value = 2000 msec

INI File Parameter Name = telFlashHookNoDialToneDelayMs

3.12.3 Inform On PBX Calling Number Update (Phone Emulating Only)

Description: When this parameter is enabled (Yes), the Media Gateway will send an informational VoIP message to the IP peer when the calling number on the circuit side changes during the call.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Media Gateway will generate this informational message.

- No = Media Gateway will not generate this informational message.

Default Value = No

INI File Parameter Name = gwInformUpdatedCallingNumber

3.12.4 Inform On No PBX CPID (Phone Emulating Only)

Description: When this parameter is enabled (Yes), the Media Gateway will generate an informational VoIP message to the IP peer if no CPID information is available for the call within the prescribed time-out.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes =Media Gateway will generate this informational message.
- No = Media Gateway will not generate this informational message.

Default Value = No

INI File Parameter Name = gwInformIfNoPbxCpid

3.12.5 Inform On No PBX CPID Time (Phone Emulating Only)

Description: Specifies the time in milliseconds that the Media Gateway will send the informational VoIP message indicating that no CPID is available for the call.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 1000 to 60000 milliseconds

Default Value = 4000 milliseconds

INI File Parameter Name = gwInformIfNoPbxCpidMs

3.12.6 ISDN Overlap Receive Minimum Digits

Description: Specifies the minimum number of called-party digits that must be received on an inbound call before the inbound call is processed. If the value is 0, then the call is processed immediately. Applicable only when the switch is using overlap sending.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 64 digits

Default Value = 1

INI File Parameter Name = isdnOverlapRcvMinDigits

3.12.7 ISDN Overlap Receive Timeout (msecs)

Description: Specifies the number of milliseconds before the call is answered when in overlap receive mode.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 0 to 65535 milliseconds

Default Value = 14000 milliseconds

INI File Parameter Name = isdnOverlapRcvToutMs

3.12.8 RFC 3960 Early Media Support

Description: Determines if the Media Gateway will support early media when requested by a SIP caller Use Agent Client (UAC).

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Gateway will support early media.
- No = Gateway will not support early media.

Default Value = Yes

INI File Parameter Name = sipEarlyMediaSupport

3.12.9 Start Port for RTP

Description: Specifies the first UDP IP port used for Realtime Transport Protocol (RTP) traffic. RTP port number assignments increment from this starting value.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values: 1024 to 65000

Default Value = 49000

INI File Parameter Name = gwRTPStartPort

3.12.10 Unauthenticated SRTP Enable

Description: This parameter can be used to force Realtime Transport Protocol (RTP) packets to be received without authentication even if an authentication algorithm has been negotiated between parties.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = RTP VOICE packets will NOT be authenticated.
- No = RTP VOICE packets will be authenticated per the negotiated algorithm.

Default Value = No

INI File Parameter Name = srtpUnAuthenticatedSRTP

3.12.11 UnEncrypted SRTCP Enable

Description: This parameter can be used to force SRTCP packets to be transmitted in an unencrypted fashion even if cipher keys have been negotiated between parties.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Transmitted RTCP CONTROL packets will NOT be encrypted despite the negotiation of cipher keys.
- No = Transmitted RTCP CONTROL packets will be encrypted per the negotiated cipher keys.

Default Value = No

INI File Parameter Name = srtpUnEncryptedSRTCP

3.12.12 UnEncrypted SRTP Enable

Description: This parameter can be used to force Secure Realtime Transport Protocol (SRTP) packets to be transmitted in an unencrypted fashion even if cipher keys have been negotiated between parties.

Note: This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Transmitted RTP VOICE packets will NOT be encrypted despite the negotiation of cipher keys.
- No = Transmitted RTP VOICE will be encrypted per the negotiated cipher keys.

Default Value = No

INI File Parameter Name = srtpUnEncryptedSRTP

3.12.13 Wait for Dial Tone after Flash Hook

Description: Indicate if the gateway should wait for dial tone after flash hook.

Notes: *1.* This parameter is only applicable to T1/E1 CAS.

2. This parameter can only be changed in the configuration file. It is not accessible through the Web interface.

Allowed Values:

- Yes = Gateway will wait for dial tone after a Flash Hook as a confirmation that the switch recognized the event.
- No = Gateway does not expect the switch to create dial tone after a Flash Hook.

Default Value = Yes

INI File Parameter Name = telDialToneExpAfterFlashHook

Working with Call Progress Tones 4

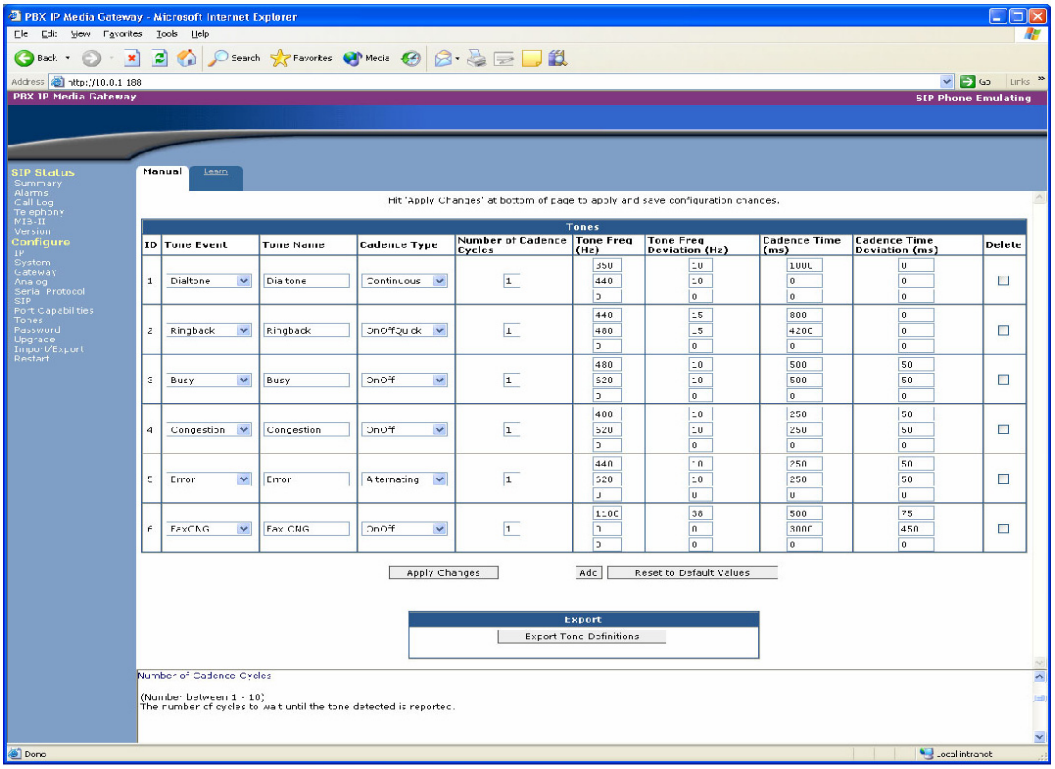
Call progress tone parameters define the characteristics (frequencies, durations, and deviations) of the tones that the Dialogic® Media Gateway detects during call progress analysis. The following sections discuss how to view, edit, learn, and validate call progress tones using a Web interface:

- Viewing and Editing Call Progress Tones 135
- Learning and Validating Call Progress Tones 136

4.1 Viewing and Editing Call Progress Tones

If call progress tone characteristics are known, call progress tones can be viewed and edited manually from the Manual Tones Web page shown in Figure 8. Or the Media Gateway can be directed to analyze and learn the characteristics of specific call progress tones as described in Section 4.2, "Learning and Validating Call Progress Tones", on page 136.

Figure 8. Manual Tones Web Page


PBX IP Media Gateway - Microsoft Internet Explorer
PBX IP Media Gateway
SIP Phone Emulating
SIP Status
Summary
Alarms
Call Log
Telephony
MGII
Version
Configure
IP
System
Gateway
Analog
Serial Protocol
SIP
Port Capabilities
Tones
Password
Upgrade
Import/Export
Restart
Manual
Learn
Hit 'Apply Changes' at bottom of page to apply and save configuration changes.
Tones
ID
Tone Event
Tone Name
Cadence Type
Number of Cadence Cycles
Tone Freq (Hz)
Tone Freq Deviation (Hz)
Cadence Time (ms)
Cadence Time Deviation (ms)
Delete
1 Dialtone Dialtone Continuous 1 350 440 0 20 20 0 1000 0 0 0 0 0 0 0 0 0
2 Ringback Ringback On/Off Quick 1 440 480 0 25 25 0 800 4200 0 0 0 0 0 0 0 0 0
3 Busy Busy On/Off 1 480 620 0 20 20 0 500 500 0 0 0 0 50 50 0 0 0
4 Congestion Congestion On/Off 1 400 620 0 20 20 0 250 250 0 0 0 0 50 50 0 0 0
5 Error Error Alternating 1 440 620 0 20 20 0 250 250 0 0 0 0 50 50 0 0 0
6 FaxCNG Fax CNG On/Off 1 1100 0 0 30 0 0 500 3000 0 0 75 450 0 0 0 0
Apply Changes
Add
Reset to Default Values
Export
Export Tone Definitions
Number of Cadence Cycles
(Number between 1 - 10)
The number of cycles to wait until the tone detected is reported.
Done
local intranet

You can also add and delete call progress tones from this Web page.

Use the following buttons on the Manual Tones Web page to manipulate call progress tones:

Apply Changes button
: Saves changes to the configuration once the parameters have been edited.

Add button
: Adds a row to the tone table, allowing a new tone definition to be added.

Reset to Default Values button
: Deletes all existing tones and restores factory default tones.

Export Tones Definitions button
: Exports the current tone definitions to an INI file. This INI file can then be imported into other Media Gateway, which will update only the call progress tones of the gateway. The tones will also be exported as part of the system configuration information.

For details on the call progress tone parameters, see Section 3.11, "Tone Parameters", on page 125.

4.2 Learning and Validating Call Progress Tones

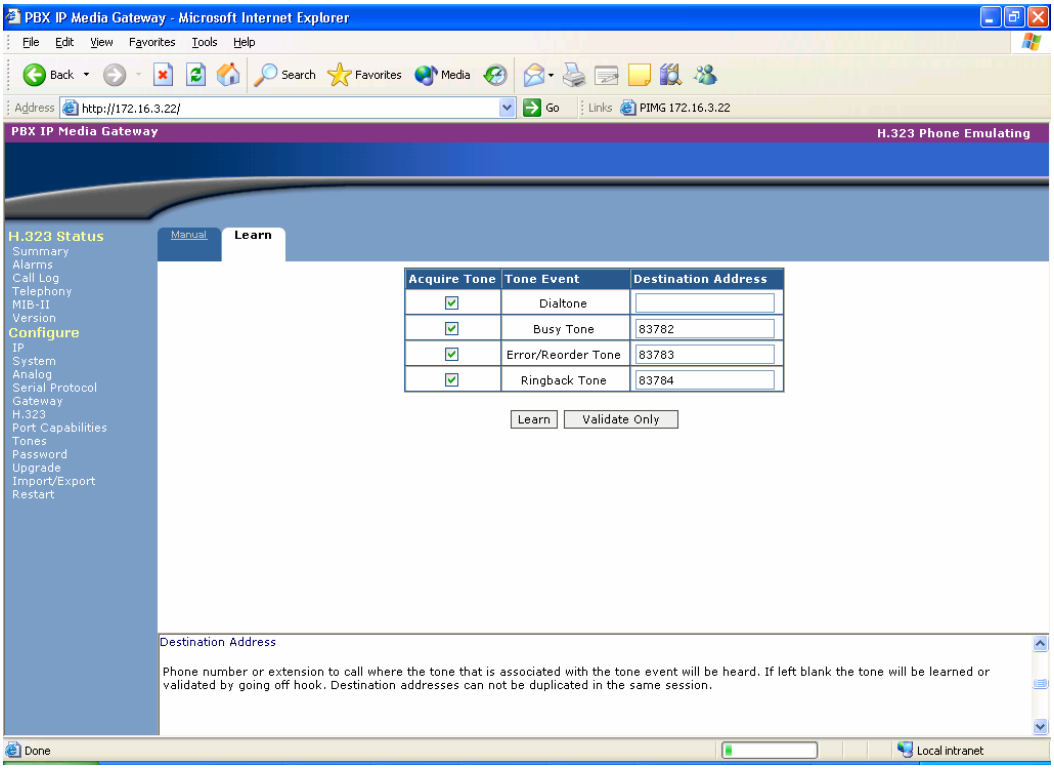
Learning and validating a call progress tone is discussed in the following topics:

- Learn Tone Web Page
- Learning the Characteristics of Unknown Call Progress Tones
- Learn Tone Progress Web Page
- Learn Tone Results Web Page
- Validating Call Progress Tones
- Validate Tone Progress Web Page
- Validate Tone Results Web Page

4.2.1 Learn Tone Web Page

The Learn Tone Web page shown in Figure 9 allows you to acquire the characteristics for new call progress tones from the telephony system or to validate the defined call progress tones.

Figure 9. Learn Tone Web Page



The Learn Tone user interface has the following components of interest:

Acquire Tone field
: A check indicates that the corresponding tone event is to be learned or validated.

Tone Event field
: Indicates one of four types: dialtone, busy, error, or ringback.

Destination Address field
: Specifies the phone number or extension that when dialed will generate the call progress tone associated with the tone event. If left blank, the tone will be learned or validated by going off hook. Destination addresses cannot be duplicated in the same session.

Note: It is recommended that you dial each of the destination addresses from a separate phone to validate that the expected call progress tones to be learned or validated are produced.

The following shows possible ways in which call progress tones can be produced:

- Dialtone – For a system dialtone, leaving the destination address blank will allow the unit to determine the dialtone by simply going off hook. For a secondary dialtone, enter the access code for the secondary dialtone in the destination address.
- Busy – Call an extension or number of a phone that is in use. **Caution:** Make sure that the extension is not forwarded on busy and that a busy tone is generated when the extension is dialed.
- Error – Call an invalid extension.

- Ringback – Call a valid extension not in use. **Caution:** Make sure that this extension is not forwarded on "no answer" and that it will ring until answered.

Learn button
: Specifies that the Media Gateway will automatically learn the tone. Used when the frequency and timing information of a call progress tone are unknown. For more information on using the Learn feature, see Section 4.2.2, "Learning the Characteristics of Unknown Call Progress Tones", on page 138.

Validate Only button
: Specifies that the Media Gateway will validate a call progress tone whose frequency and timing information are known. For more information on using the Validate only feature, see Section 4.2.5, "Validating Call Progress Tones", on page 144.

4.2.2 Learning the Characteristics of Unknown Call Progress Tones

If the frequency and timing information of a call progress tone is unknown, use the Learn button to have the Media Gateway automatically learn the tone. This is accomplished by selecting the tone, entering a destination address where the desired call progress tone can be heard and selecting Learn. The unit will then call each destination address and analyze the tones on the line. The result will be a tone definition for the specified tone event that can be added to the tone table.

Follow these steps to learn the timing and frequency characteristics of unknown call progress tones.

1. For each call progress tone that is to be learned, determine how the tone can be generated by the system.

2. Verify that the system will generate the desired call progress tone by dialing that destination address from an extension and listening to verify the tone.

3. Navigate to the Learn Tone Web page (see Figure 9), select the desired tones, and enter the destination addresses for the tones to be learned.

4. Click on the Learn button.

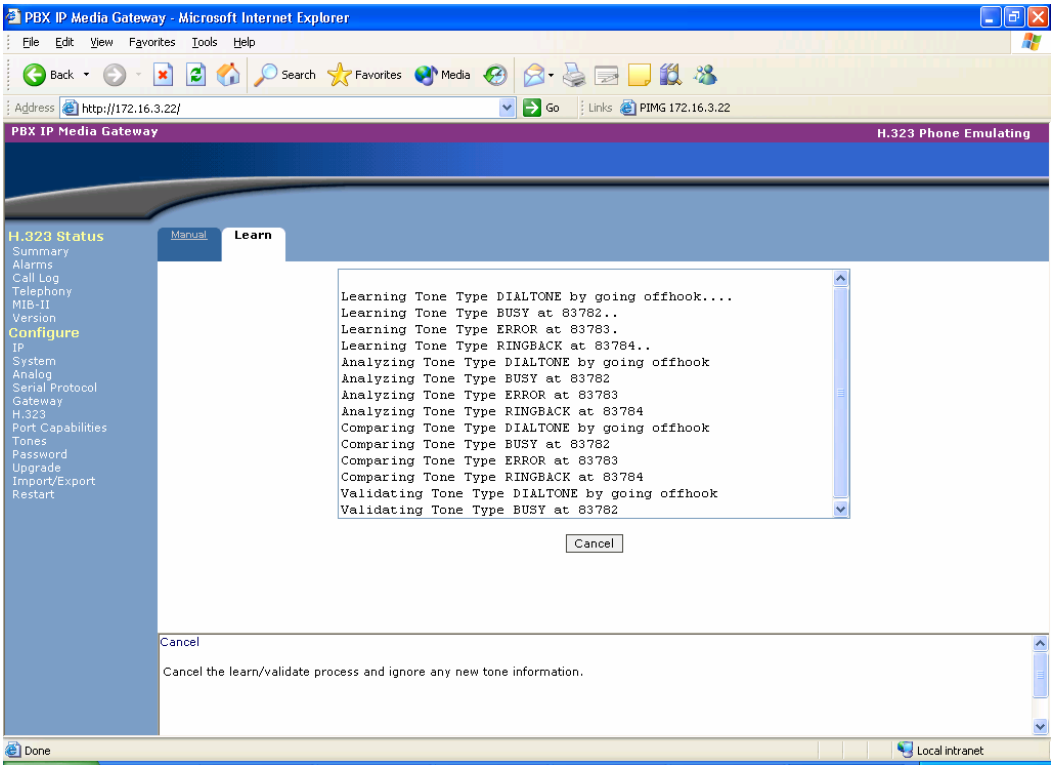
5. Wait for the Learn Tone Results Web page to be displayed (see Figure 11 and Figure 12). A progress page (see Figure 10) is displayed showing progress messages as the system proceeds.

6. Select the newly learned tones to be added to the tone definition table and click on the Apply button.

4.2.3 Learn Tone Progress Web Page

Once the Learn button has been selected, the Media Gateway will start the process of acquiring the call progress tone characteristics. A Learn Tone Progress page, shown in Figure 10, will automatically appear while the Gateway is acquiring the tones and progress messages will be posted.

Figure 10. Learn Tone Progress Web Page



Progress messages are described as follows:

Learning
The Gateway has gone off hook and dialed the destination address. It is collecting raw data on the call progress tone.

Analyzing
The Gateway is analyzing the raw data to extract the tone characteristics.

Comparing
The Gateway is comparing the new tones to existing tones and other new tones for possible conflicts.

Validating
The Gateway is redialing the destination addresses and testing to see if the new call progress tone definitions can detect the tones.

The Cancel button cancels the current session and returns to the Learn Tone Web page.

4.2.4 Learn Tone Results Web Page

When the Media Gateway has finished learning the tones, a results page, shown in Figure 11 and Figure 12, is displayed.

Figure 11. Learn Tone Results Web Page (Top)

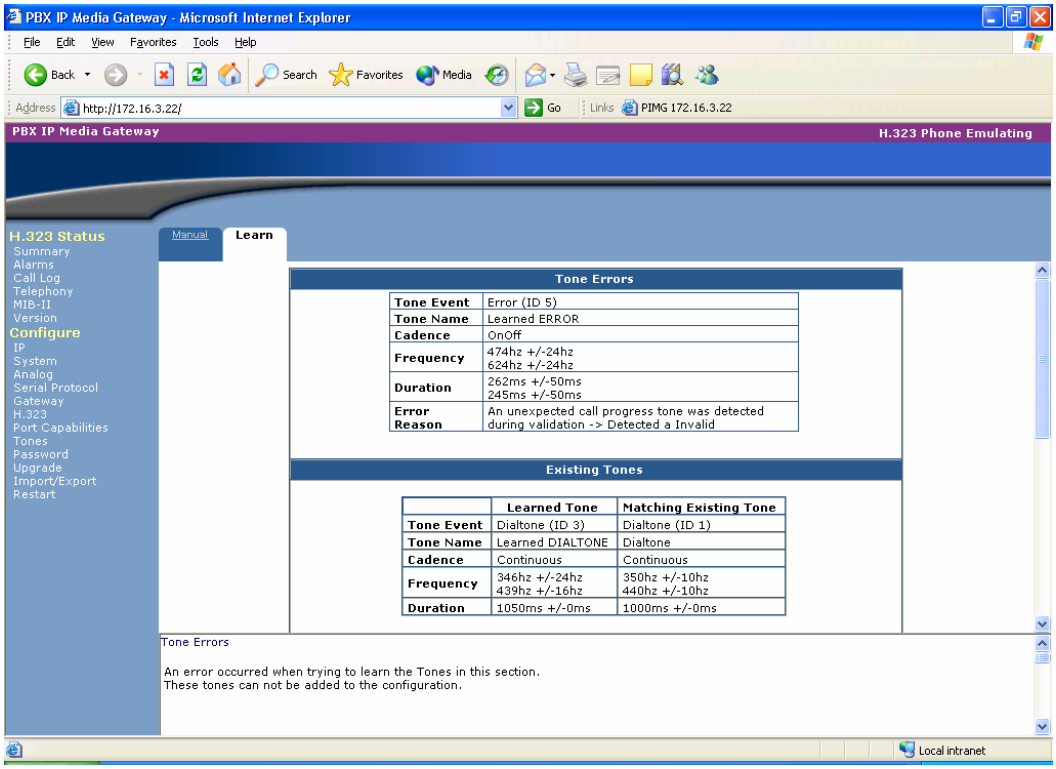
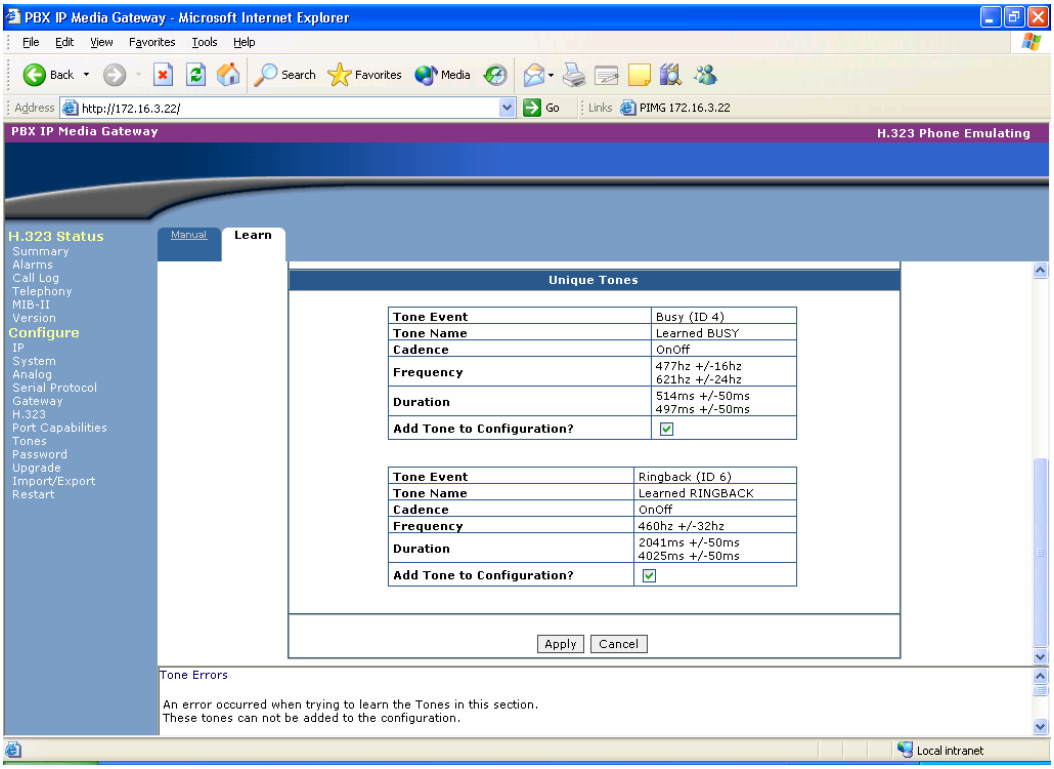


Figure 12. Learn Tone Results Web Page (Bottom)



The learned tones are grouped into the following possible tone results on the Web page:

- Tone Errors
- Existing Tones
- Unique Tones
- Conflicting Tones

After selecting the tones to be added to the tone definition table, the Apply button adds the selected tones to the tone definition table. The Manual Tones Web page discussed in Section 4.1, "Viewing and Editing Call Progress Tones", on page 135 can then be used to view the newly learned tones.

Tone Errors

An error occurred when trying to learn the tones listed in this section. These tones cannot be added to the configuration due to the error encountered. Possible reasons for error as well as possible action include the following:

No data

The Gateway dialed the destination address but there was silence on the line and no data could be collected.

Action: Verify the destination address.

Not enough data

The Gateway dialed the destination address but the data collected was not enough to extract tone characteristics.

Action: Verify the destination address and ensure that the destination does not forward to another extension or that the destination is not answered.

Failed Validation unexpected CP tone

The Gateway dialed the destination address and was able to extract tone characteristics, but when the Gateway dialed the destination address to validate the newly learned tone, an unexpected call progress (CP) tone was detected. The detected CP tone is listed in the error message. This detected CP tone did not match the newly learned tone or any expected conflicts. This detected CP tone may be a tone that already exists in the tone definition table or another newly learned tone.

Action: Using the listed tone characteristics and the unexpected tone event that was detected, it may be possible to compare tones and determine which of the tones was actually detected. If the tone definitions only overlap in the time deviation, then it may be possible to manually add the tone and adjust the time deviation of the new tone and the conflicting tone to give two unique tones. If the two tones are the same, then only one tone can be used.

Failed Validation Timeout

The Gateway dialed the destination address and was able to extract tone characteristics, but when the Gateway dialed the destination address to validate the newly learned tone, no CP tone was detected.

Action: Validate the destination address.

Existing Tones

The tones listed in this section were successfully learned but found to match tones already in the configuration. Since these tones are already in the tone definition table no further action is needed.

Unique Tones

The tones listed in this section were successfully learned and validated without error or conflicts. If the "Add tone to Configuration?" check box is checked, these tones will be added to the tone definition table when you select the Apply Changes button.

Conflicting Tones

The tones listed in this section were successfully learned but were found to conflict with other tones. You can select only one of the conflicting tones to be added to or kept in the configuration. If the tone that conflicts with the selected tone is already in the tone definition table, it will be deleted when you select the Apply Changes button. If the conflicting tone is another newly learned tone, it will be ignored.

Note: If the conflicting tones do not overlap by much and it is desired to keep both conflicting tones, you can print the results page or write down the tone characteristics for the conflicting tone and add it to the tone definition table manually. You will need to manually adjust the tone characteristics of the conflicting tones to remove the conflict.

Learn Tone Issues and Possible Solutions

When comparing tone definitions, both the frequency and timing need to be considered.

To compare frequencies, both the frequency itself and the frequency deviation are needed. When a frequency is detected on the line, it is compared to the bandwidth of the tone definition. The bandwidth is specified as all of the frequencies from the specified frequency minus the frequency deviation to the specified frequency plus the frequency deviation. For example, if the frequency is specified as 400 Hz and the frequency deviation is specified as 50 Hz then the bandwidth is 350 Hz to 450 Hz.

The frequencies of tone definitions are considered to match if their bandwidths overlap. For example, if tone 1 has frequency 400 Hz with frequency deviation of 50 Hz, and tone 2 has frequency 460 Hz with frequency deviation of 20 Hz, then these frequencies would match. The bandwidth of tone 1 is 350 through 450 Hz, and the bandwidth of tone 2 is 440 through 480 Hz.

Some common learn tone issues and possible solutions are provided here:

An On/Off tone is detected while validating a stutter dialtone

A stutter dial tone is a dial tone that has a short period of 2 or 3 cycles of On/Off cadence before the dialtone is on continuously. In most cases the stutter dial tone will not be an issue as the dial tone will have distinct frequencies. In those rare cases where the dial tone shares the same frequencies as On/Off tones, it may be possible that the dialtone will experience validation errors. This may happen if the dial tone stutter period matches the on and off timing of a tone with the same frequencies. Since there is no conflict between Continuous tones and On/Off tones, the conflict between On/Off tones and the stutter part of the stutter dial tone will not be detected automatically.

Possible solution: Adjust the timing of the On/Off tone, or set the number of cadence cycles for the On/Off tone to a value greater than 1.

An On/Off tone with double cadence conflicts with another On/Off tone

A double cadence is an On/Off cadence with two different on and two different off periods. For example, a ringback tone may have the following cadence: on for 500 msec, off for 500 msec, on for 1500 msec, off for 3500 msec. This cadence will automatically be learned as an On/Off tone with time on as 1000 msec with deviation of 500 msec and Off time of 2000 with deviation of 1500 msec. Since this timing has such wide deviation it may conflict with other tones with the same frequency. For example, if there is a busy tone with the same frequency and the cadence of on for 500 msec, off for 500 msec, then this busy will conflict with the ringback definition.

Possible solution: To prevent the busy tone event during ringback, increase the number of cadence cycles for the busy tone to 2. To prevent ringback event during busy, change the cadence type of the ringback to OnOffQuick. Set the first time value to a value that is greater than the time on for the busy tone, but less than the larger on time of the ringback. For this example, 1000 msec would be a good choice. Set the second time value to a value larger than the longest off time for the ringback. For this example, 3600 msec would be a good choice.

Note: Changing the cadence type to OnOffQuick may not work if there is already a continuous type or OnOffQuick type with matching frequencies. In this case, the Gateway will not be able to reliably detect the tone with a double cadence.

4.2.5 Validating Call Progress Tones

If the timing and frequency information is available and the desire is to only validate the information, then use the Validate Only button. The Media Gateway will dial a destination address where the call progress tone to be tested can be heard. The unit will then call each destination address and determine if the specified call progress tone is detected.

The validation process is useful when there was an error in learning new tones and some tone characteristics have been manually adjusted to remove conflicts. The Validate Only button can be used to validate the changes.

Follow these steps to validate the timing and frequency characteristics of call progress tones.

1. For each call progress tone that is to be validated, determine how that tone can be generated by the system.

2. Verify that the system will generate the call progress tone that is desired by dialing that destination address from an extension and listening to verify the tone.

3. Navigate to the Learn Tone Web page (see Figure 9) and enter the destination addresses for the tones to be learned.

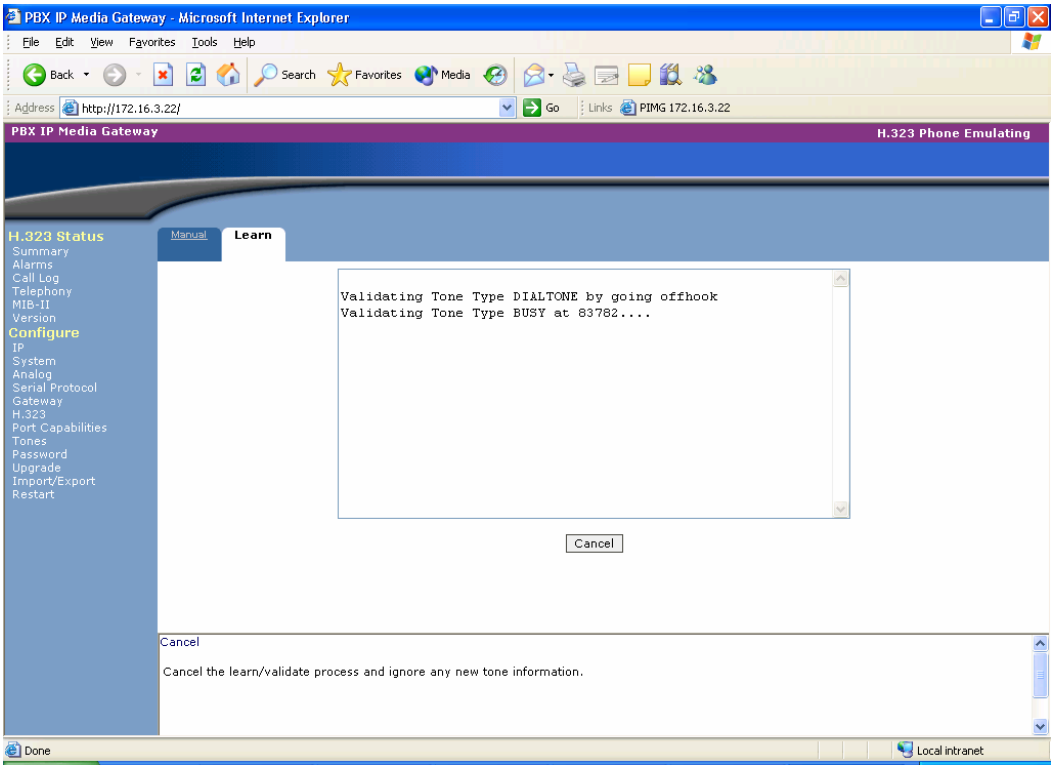
4. Click on the Validate Only button.

5. Wait for the Validate Tone Results page to be displayed (see Figure 14 and Figure 15). A progress page will be displayed (see Figure 13) showing progress messages as the system proceeds.

4.2.6 Validate Tone Progress Web Page

Once the Validate Only button has been selected, the Media Gateway will start the process of validating the call progress tone. A progress page, shown in Figure 13, will automatically appear while the Gateway is validating the tones and progress messages will be posted.

Figure 13. Validate Tone Progress Web Page



The message, Validating, means that the Gateway is redialing the destination addresses and testing to see if the call progress tone definitions can detect the tones.

The Cancel button cancels the current session and returns to the Learn Tone Web page.

4.2.7 Validate Tone Results Web Page

When the Media Gateway has finished validating the tones, a results page, shown in Figure 14 and Figure 15, will be displayed.

Figure 14. Validate Tone Results Web Page (Top)

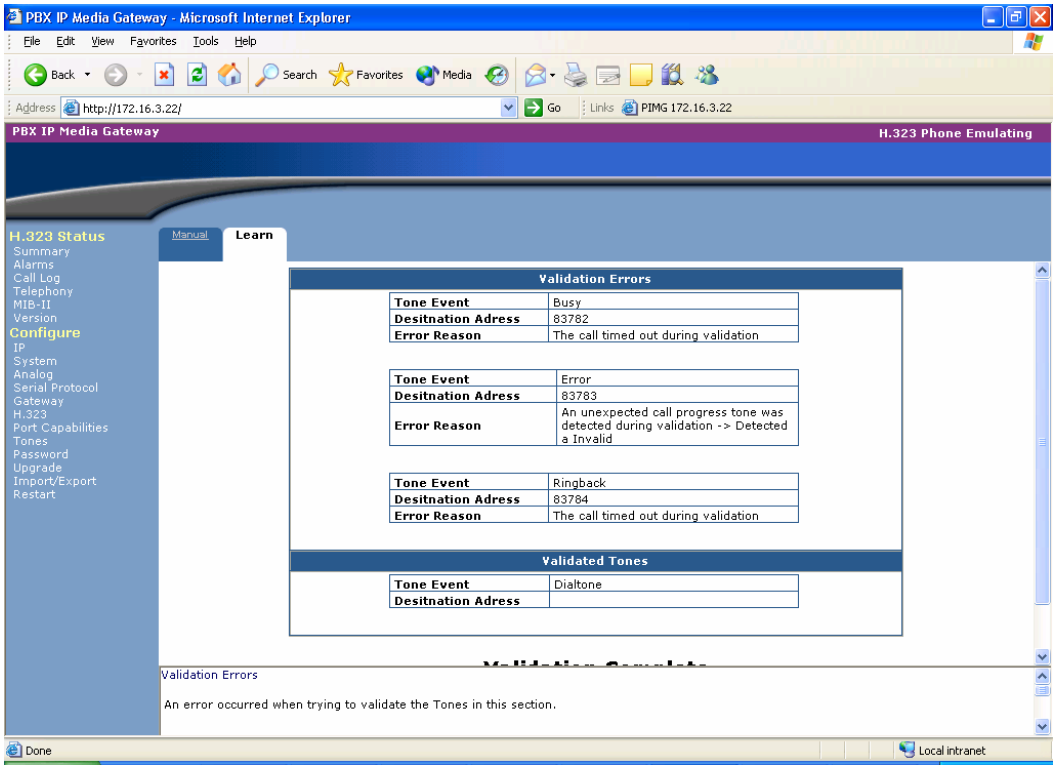
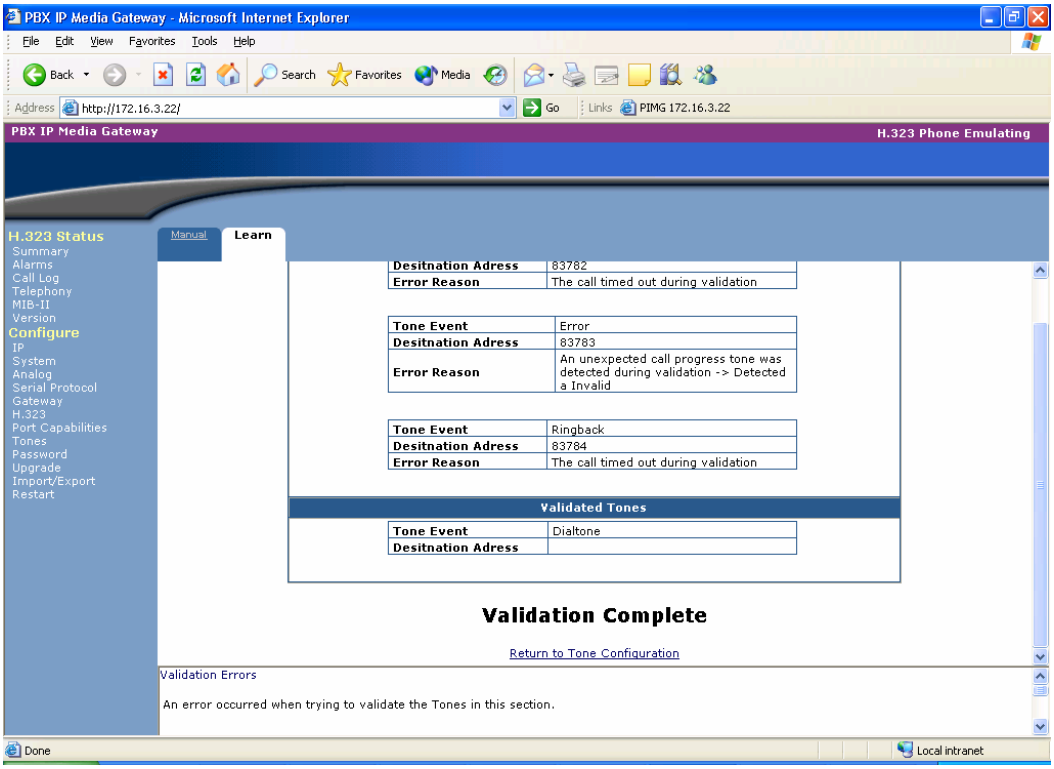


Figure 15. Validate Tone Results Web Page (Bottom)



The validated tones are grouped into the following tone results:

- Validation Errors
- Validated Tones

After viewing the validation results, you can return to the Manual Tones Web Page (see Figure 8) by clicking on the Return to Tone Configuration link.

Validation Errors

An error occurred when trying to validate the tones listed in this section. These tones cannot be added to the configuration due to the error encountered. Possible reasons for error include:

Failed Validation unexpected CP tone

When the Gateway dialed the destination address to validate the tone, an unexpected call progress (CP) tone was detected. The detected CP tone is listed in the error message.

Failed Validation Timeout

When the Gateway dialed the destination address to validate the tone, no CP tone was detected.

Validated Tones

The tones listed in this section were successfully validated.

Dial Plan 5

Information about dial plan and how it is supported by the Dialogic® Media Gateway is described in the following sections:

- Dial Plan Overview 149
- CPID (Number) Matching 152
- Channel Pools 152
- CPID (Number) Manipulation 154
- Inbound VoIP Call Routing 156
- Inbound TDM Call Routing 157
- Web Configuration Notes 158
- Offline Testing 159
- Implementation Example 161

5.1 Dial Plan Overview

The dial plan describes a set of rules used to define the characteristics of a call routed through the gateway. The primary characteristics include the destination address and the CPID (call party identification) information. The dial plan affects calls originating from the VoIP side, and calls originating from the TDM (T1, E1, analog, etc.) side.

5.1.1 VoIP to TDM Calls

Calls originating from the VoIP interface have two associated URLs. One contains the originating address and a calling number. The other contains a destination address and a called number. The dial plan uses this information to determine a final TDM destination for the call. Let's say a call occurs as follows:

From: 101@172.16.3.131
To: 8675309@172.16.3.200

The user at address 172.16.2.131 is attempting to connect to the gateway at 172.16.3.200. The calling number is 101, and the called number is 8675309. Figure 16 below shows how this information enters the dial plan control.

Figure 16. VoIP to TDM calls

The dial plan configuration will determine what calling and called numbers are sent to the TDM device, as well as the physical TDM destination.

5.1.2 TDM to VoIP Calls

Calls originating from the TDM interface have a physical source, along with calling, called, and redirect number information. The dial plan uses a combination of the physical source, calling, called, and redirect numbers to determine the URL of the destination call. Take the case where a call occurs as follows on a T1 interface:

Physical Source: Interface 2, Channel 6
Calling Number: 5402
Called Number: 95551212

A call is being received on the 6th channel of physical interface 2. The calling number (the caller) is 5402, and the number called (dialed) is 95551212. Figure 17 below shows how this information enters the dial plan control.

Figure 17. TDM to VoIP calls

The dial plan configuration will determine the calling number, called number, redirect number, and VoIP address of the destination call. However, only the calling and called numbers can be used to route the call (i.e. the redirect number is not considered for routing purposes).

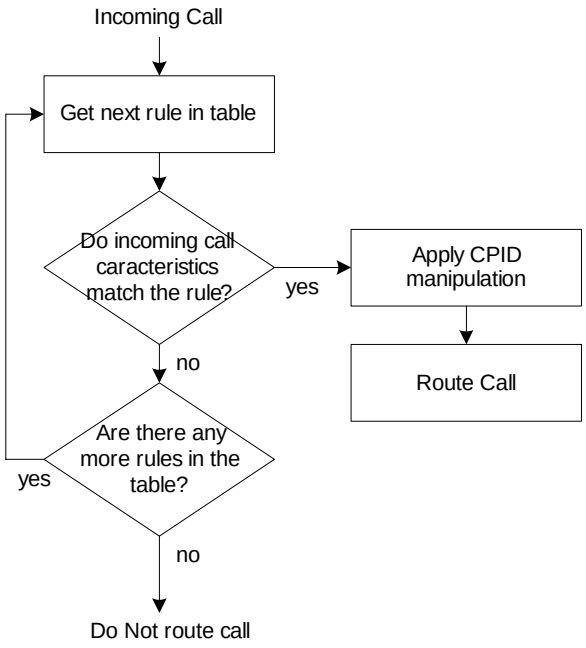
5.1.3 Determining the Call Destination

The dial plan uses four tables configured by the user to determine the final destination.

- The "Channel Pool" table creates a logical grouping of interfaces and channels that define the physical interface for the TDM side of the call.
- The "CPID Manipulation" table is used to create a set of formulae used to establish the calling, called, and redirect numbers for the destination call.
- The "Inbound VoIP Call Routing" table provides a set of rules used to determine the route of incoming VoIP calls.
- The "Inbound TDM Call Routing" table provides a set of rules used to determine the route of incoming TDM calls.

As a call enters the gateway, the incoming characteristics of the call are matched against the dial plan rules in the appropriate routing table from top down. If a match is found, the CPID information is updated based on the CPID manipulation entry in the table assigned to the matched rule. The call is then sent to the destination assigned in the table. Figure 18 below shows call routing flow for dial plan.

Figure 18. Dial Plan Call Routing Flow



If no match is found, the call is not routed by the dial plan.

5.2 CPID (Number) Matching

Part of the criteria for matching the characteristics of an incoming call to a rule in one of the dial plan routing tables is number matching. The incoming calling or called number is tested against a match rule defined in the routing table. If the match fails, this routing rule table entry will fail.

The formula for number matching gets entered via the dial plan configuration in the "Inbound VoIP Call Routing" Web page and the "Inbound TDM Call Routing" Web page. Table 2 shows the syntax used for number matching. The example shows an attempted match with the following number… 7168675309

Table 2. Syntax Used for Number Matching

Token	Description	Example	Example Result
0 1 2 3 4 5 6 7 8 9	Digit	7168675309 7168675308	Match No Match
[digit, digit-digit]	Match any single digit in range	716[2,5-9]675309 716[2,5-7]675309	Match No Match
x	Match any single digit	716xxx5309 716xxxx5309	Match No Match
.	Match any number of *ending* digits	x16. x15.	Match No Match
*	Match all	xxxxxxxxxx xxxxxxxxx	Match No Match

5.3 Channel Pools

5.3.1 Description

A channel pool is a logical grouping of interfaces and channels that define the physical interface for the TDM side of the call. All incoming TDM calls arrive on a physical interface and a channel. A channel pool is a way of identifying the interface / channel combination. Outgoing calls are made to a physical interface and a channel. For some calls, it may be necessary to select the specific interface and channel from a group of possible options. A channel pool defines how this is selected.

5.3.2 TDM Interface Types

The gateway supports multiple types of TDM interfaces. The number of physical interfaces and channels supported vary. Table 3 shows the possibilities:

Table 3. TDM Interface Types

Gateway Type	Maximum number of supported Interfaces	Number of supported Channels per interface
Analog	8	1
Digital (PBX)	8	1
T1 - CAS	4	24
T1 - ISDN	4	23
E1	4	30

5.3.3 Channel Selection Mode

It is often desirable to select a method for choosing which channel within a channel pool is used to place an **outgoing** TDM call. The channel pool configuration uses a channel selection mode configuration option for this. The valid options are:

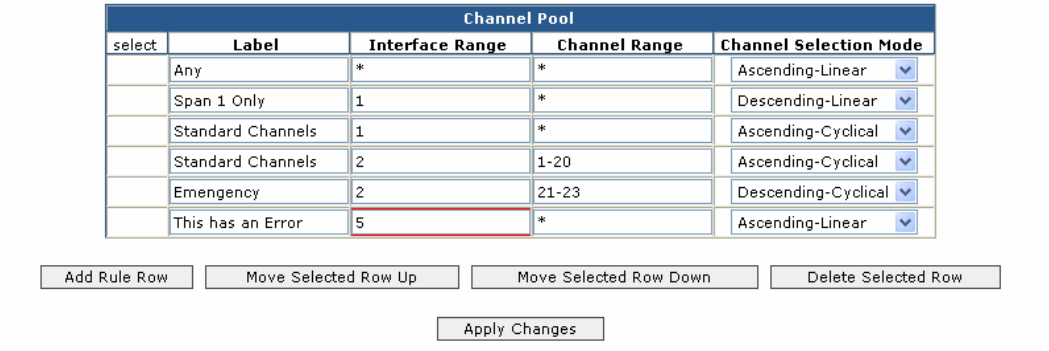
- Ascending Linear - The first (lowest logical) available channel is chosen to place the call.
- Ascending Cyclical - Channels are chosen upward sequentially from call to call. The selection wraps back to the first channel when the limit is reached.
- Descending Linear - The last (highest logical) channel is chosen to place the call.
- Descending Cyclical - Channels are chosen downward sequentially from call to call. The selection wraps back to the last channel when the limit is reached.

Note: *Channel Selection Mode* applies to **outgoing calls** only.

5.3.4 Web Configuration Interface

Figure 19 is a sample screen shot of the channel pool configuration Web page:

Figure 19. Channel Pool Configuration Web Page



Select - Click anywhere in this column to select an entry. A selected row may be moved up or down in the table, or deleted.

Label - Label for this entry. This label is referenced in the "Inbound VoIP Routing" and the "Inbound TDM Routing" configuration pages.

Interface Range - The 1-based physical interface(s) assigned to this channel pool. Numbers can be entered in the form #,# to select individual interfaces, or #-# to select a range of interfaces, or a combination of both. For example, 2,5-7 would use interfaces 2,5,6, and 7. The '*' character represents all available interfaces.

Channel Range - The 1-based logical channel(s) within the interface(s) assigned to this channel pool. Numbers can be entered in the form #,# to select individual channels, or #-# to select a range of channels, or a combination of both. For example, 2,5-7 would use channels 2,5,6, and 7. The '*' character represents all available channels.

Channel Selection Mode - The channel selection mode used for outgoing TDM calls. This is only relevant when this channel pool was used in the "Outgoing TDM Information" area in the "Inbound VoIP Routing" configuration page.

See Section 5.7, “Web Configuration Notes”, on page 158 for more details on using the Web interface.

5.4 CPID (Number) Manipulation

CPID manipulation defines rules for how calling, called, and redirect numbers appear in the destination call. Typically, the calling and called numbers from the incoming call are simply passed through to the outgoing call. However, there are cases where it is desirable to alter either or both of these numbers. A "CPID Manipulation" rule is applied to all calls routed via the dial plan. Each of the rules applies a formula using the incoming CPID information as inputs to determine the outgoing call's CPID.

5.4.1 Rules Syntax

Table 4 shows the syntax used for CPID Manipulation. The example data is an incoming VoIP call:

From: 7168675309@172.16.3.15
To: 5551212@10.20.34.78

Table 4. Syntax Used for CPID Manipulation

Rule Syntax	Description	Example	Example Result
S	Source (calling) number	S	7168675309
D	Destination (called) number	D	5551212
R	Redirection number	R	

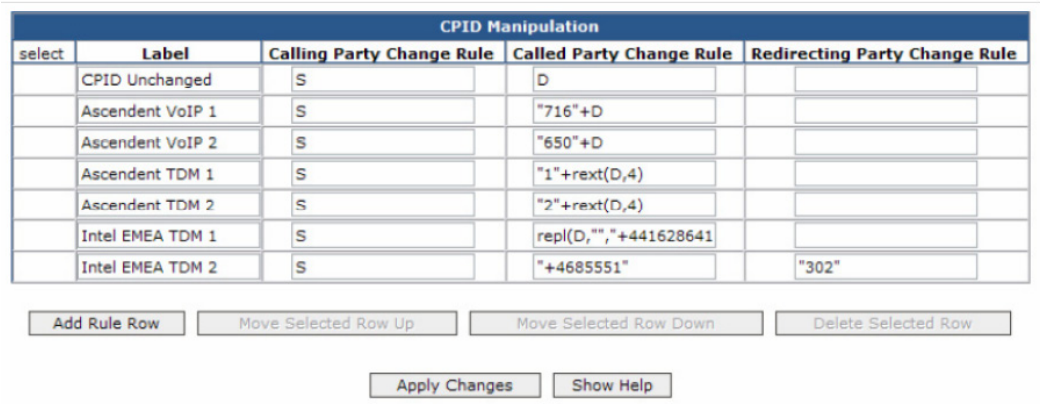
Table 4. Syntax Used for CPID Manipulation (Continued)

Rule Syntax	Description	Example	Example Result
I	Inbound VoIP address	I	172.16.3.15
" "	Takes what's in quotes as literal	"353"	353
+	Concatenate	"800" + D	8005551212
lext(str, n)	Extract ***n*** characters from left of ***str***	lext(S, 3)	716
rext(str, n)	Extract ***n*** characters from right of ***str***	rext(S, 4)	5309
lrem(str, n)	Remove ***n*** characters from left of ***str***	lrem(S, 3)	8675309
rrem(str, n)	Remove ***n*** characters from right of ***str***	rrem(D, 4)	555
mext(str, pos, n)	Extract ***n*** characters from ***str*** starting ***pos*** digits from left	mext(S, 5, 2)	53
repl(str, old, new)	Find 1st occurrence of ***old*** in ***str*** and replace with ***new***	repl(D, "12", "46")	5554612

5.4.2 Web Configuration Interface

Figure 20 is a screen shot of the CPID manipulation configuration Web page:

Figure 20. CPID Manipulation Configuration Web Page



Select - Click anywhere in this column to select an entry. A selected row may be moved up or down in the table, or deleted.

Label - Label for this entry. This label is referenced in the "Inbound VoIP Routing" and the "Inbound TDM Routing" configuration pages.

Calling Party Change Rule - The formula as specified in the section above that determines the calling party number of the outgoing call.

Called Party Change Rule - The formula as specified in the section above that determines the called party number of the outgoing call.

Redirecting Party Change Rule - The formula as specified in the section above that determines the redirecting party number of the outgoing call.

See Section 5.7, "Web Configuration Notes", on page 158 for more details on using the Web interface.

5.5 Inbound VoIP Call Routing

The Inbound VoIP Call Routing table is the main table used to route calls originating from the VoIP interface. When a call is received, the incoming call characteristics are compared against the values entered in the first row of the table. If a match is found, an action is taken. If no match is found, the next row in the table is checked. Rows will be tested until a match is found, or there are no more entries in the table.

The VoIP address, calling number, and called number **all** need to match the incoming call for the test to pass. Once a match occurs, the CPID will be modified based on the CPID manipulation rule selected in the table. Next, the outgoing call will be made out the TDM interface using the channel pool selected.

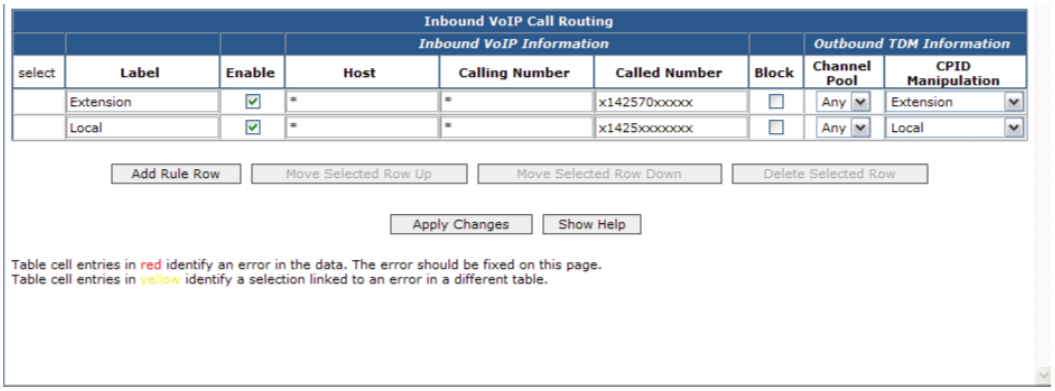
If no rows in the table successfully match the incoming call, the call is not routed.

Note: If a server list is configured, the call will still be routed via the server list.

5.5.1 Web Configuration Interface

Figure 21 is a screen shot of the Inbound VoIP Call Routing configuration Web Page:

Figure 21. Inbound VoIP Call Routing Configuration Web Page



Select - Click anywhere in this column to select an entry. A selected row may be moved up or down in the table, or deleted.

Label - Label for this entry. This label is used only to help identify the rule to the administrator. It is not used by any other tables in the dial plan configuration.

Enable - Check this box to enable the rule. If this box is not checked, this rule is not checked for incoming calls.

Host - The string to match the host address of the incoming call. This value is either an IP address in the form xxx.xxx.xxx.xxx to match a specific address, or an '*' to match any IP address.

Calling Number - The string to match the calling number of the incoming call. The format for this string is defined in the section titled "CPID (Number) Matching".

Called Number - The string to match the called number (dialed number) of the incoming call. The format for this string is defined in the section titled "CPID (Number) Matching".

Block - If this box is checked and an incoming call matches the rule, the incoming call is blocked. No outgoing call will be made. No more rules in the table will be tested.

Channel Pool - The label of an entry in the "Channel Pool" table used to specify the physical interface and channel of the outgoing TDM call.

CPID Manipulation - The label of an entry in the "CPID Manipulation" table used to create the calling and called party numbers of the outgoing TDM call.

5.6 Inbound TDM Call Routing

The Inbound TDM Call Routing table is the main table used to route calls originating from the TDM interface. When a call is received, the incoming call characteristics are compared against the values entered in the first row of the table. If a match is found, an action is taken. If no match is found, the next row in the table is checked. Rows will be tested until a match is found, or there are no more entries in the table.

For a test to pass, the channel of the incoming call must reside in the specified channel pool. The calling and called number must also match the rule entered in the table. Once a match occurs, the CPID will be modified based on the CPID manipulation rule selected in the table. Next, the outgoing call will be made to the VoIP address specified in this table.

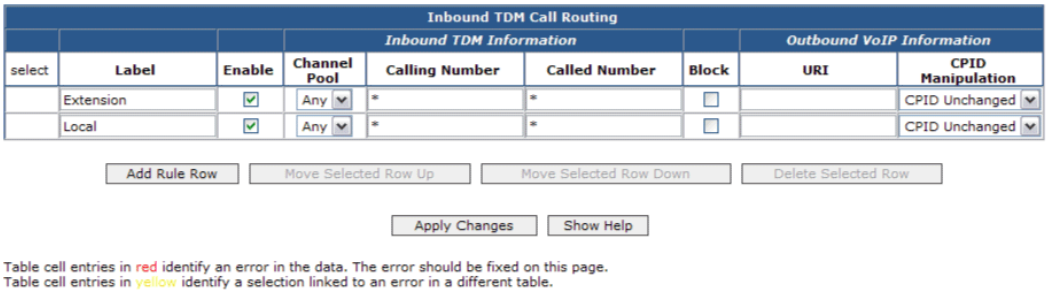
If no rows in the table successfully match the incoming call, the call is not routed.

Note: If a server list is configured, the call will still be routed via the server list.

5.6.1 Web Configuration Interface

Figure 22 is a screen shot of the Inbound TDM Call Routing configuration Web Page:

Figure 22. Inbound TDM Call Routing Configuration Web Page



Select - Click anywhere in this column to select an entry. A selected row may be moved up or down in the table, or deleted.

Label - Label for this entry. This label is used only to help identify the rule to the administrator. It is not used by any other tables in the dial plan configuration.

Enable - Check this box to enable the rule. If this box is not checked, this rule is not checked for incoming calls.

Channel Pool - The label of an entry in the "Channel Pool" table used to specify the physical interface on which the incoming call must reside.

Calling Number - The string to match the calling number of the incoming call (the "From" field of the URL). The format for this string is defined in the section titled "CPID (Number) Matching".

Called Number - The string to match the called number of the incoming call (the "To" field of the URL). The format for this string is defined in the section titled "CPID (Number) Matching".

Block - If this box is checked for a matched rule, the incoming call is blocked. No outgoing call will be made. No more rules in the table will be tested.

URI - The string to specify the URI of the outbound call.

CPID Manipulation - The label of an entry in the "CPID Manipulation" table used to create the calling and called party numbers of the outgoing TDM call.

5.7 Web Configuration Notes

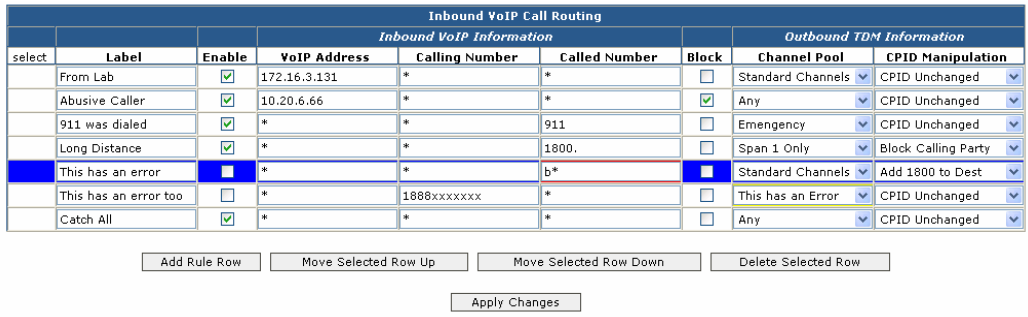
The Web interface is used for configuring all 4 tables of the dial plan. New rules can be added, existing rules can be modified, and existing rules can be deleted.

- To add a rule, click on the "Add Rule Row" button. A rule with default values will exist at the bottom of the table. Go in and change these to the desired values.

- To change a rule, edit any of the fields in a row.
- To delete a rule, click the row on the left side in the "select" area. The background of the row will change color as in the example below. Once the row has been selected, click the "Delete Selected Row" button to delete the row.
- Since rules are processed top down, it may be necessary to move a rule either up or down in the table. To do this, select the row, and then press one of the "Move Selected Row" buttons.

Note: Changes will not be permanent until the "Apply Changes" button is clicked.

Figure 23. Example of Configuration Web Page



- If a table entry is highlighted in red, there is an error in that entry. In the case above, the 'b*' is not a valid called number match rule.
- If a table entry is highlighted in yellow, there is an error in the rule that the highlighted entry references. In the case above, there is an error in the channel pool table entry that has the label "This has an Error". The error must be fixed in the corresponding table.
- Rules with an error anywhere in the row are treated as *disabled*, and are not processed.

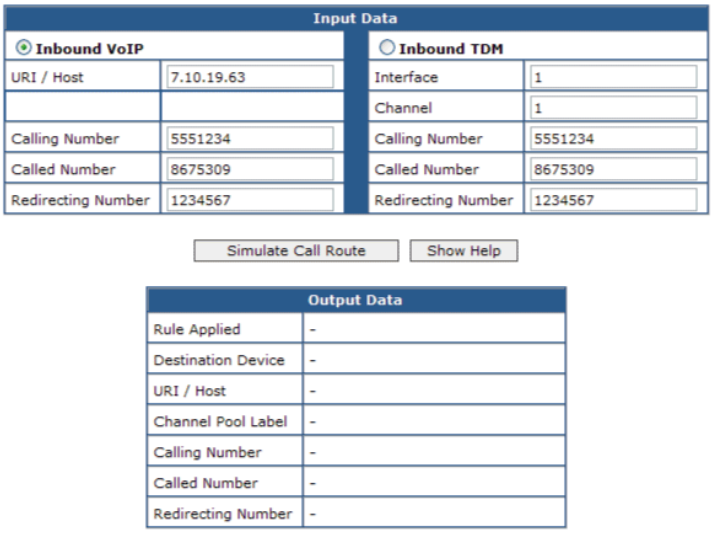
5.8 Offline Testing

An offline facility exists to test the dial plan tables without receiving actual calls. Use the "Routing Test" page for this feature.

Two tables exist in this page. The "Input Data" table is used to take simulated call characteristics from an incoming call. First, the call direction is selected, and then the call information is entered. The "Output Data" table shows the result that would occur if the simulated incoming call passed through the dial plan.

The fields for simulated data match the routing tables exactly with one exception. On inbound TDM calls, enter the interface and channel instead of the channel pool label. This helps test the channel pool configuration as well as the routing configuration.

Figure 24. Inbound VOIP Input and Output Data

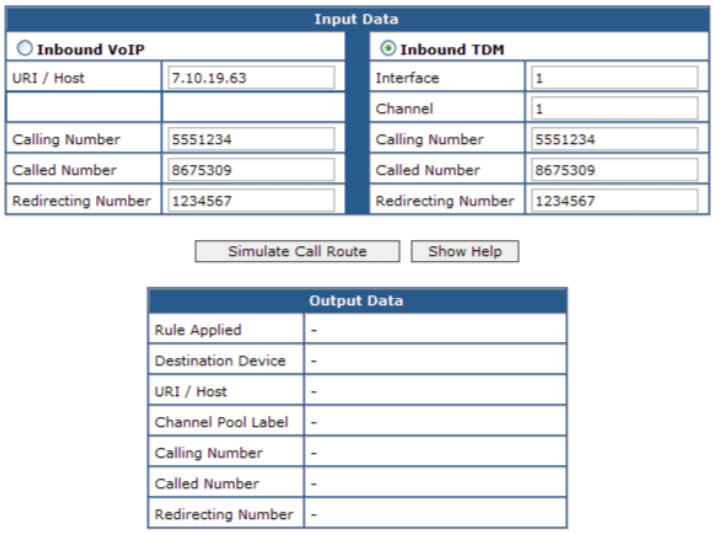


Input Data			
◉ Inbound VoIP		○ Inbound TDM	
URI / Host	7.10.19.63	Interface	1
		Channel	1
Calling Number	5551234	Calling Number	5551234
Called Number	8675309	Called Number	8675309
Redirecting Number	1234567	Redirecting Number	1234567

Simulate Call Route | Show Help

Output Data	
Rule Applied	-
Destination Device	-
URI / Host	-
Channel Pool Label	-
Calling Number	-
Called Number	-
Redirecting Number	-

Figure 25. Inbound TDM Input and Output Data



Input Data			
○ Inbound VoIP		◉ Inbound TDM	
URI / Host	7.10.19.63	Interface	1
		Channel	1
Calling Number	5551234	Calling Number	5551234
Called Number	8675309	Called Number	8675309
Redirecting Number	1234567	Redirecting Number	1234567

Simulate Call Route | Show Help

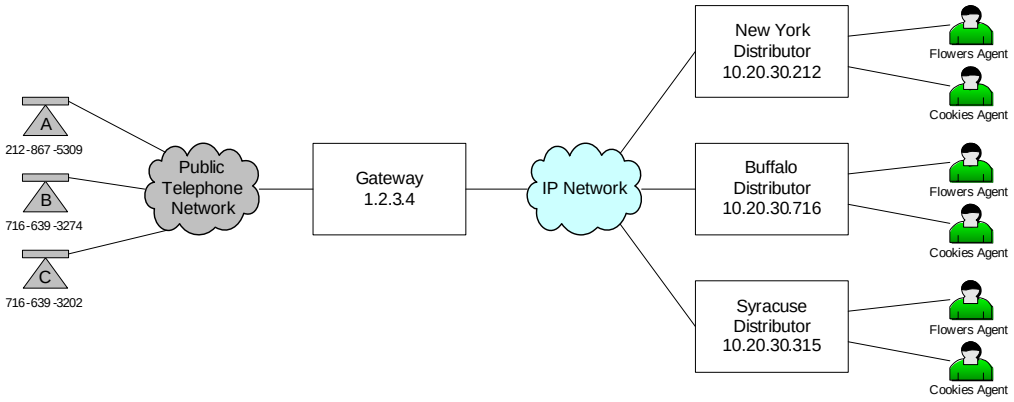
Output Data	
Rule Applied	-
Destination Device	-
URI / Host	-
Channel Pool Label	-
Calling Number	-
Called Number	-
Redirecting Number	-

5.9 Implementation Example

5.9.1 Route an Incoming TDM Call based on the Calling and Called Number

Figure 25 represents a real world scenario where a gateway would be used to route an incoming TDM call to a different VoIP endpoint based on the calling and called numbers.

Figure 26. Route an Incoming TDM Call to a Different VoIP Endpoint

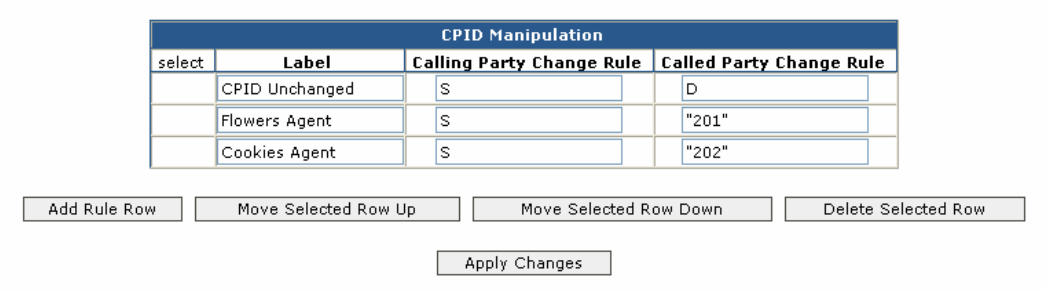


Here, we want callers from each area code directed to a specific distributor on the VoIP network. Each VoIP endpoint will have an agent assigned to an individual function. For example, if a customer from the Syracuse area wants to order cookies, the "cookies agent" at the Syracuse Distributor's VoIP address receives the call.

The implementation of this requires table entries in the TDM to VoIP routing page, and the CPID manipulation page. The default Channel Pool (Any) entry can be used.

1. **Set up the CPID manipulation rules.** When the called number is 1-800-FLOWERS (18003569377), change the outgoing VoIP called number to the extension of the flowers agent, say 201. When the called number is 1-800-COOKIES (18002665437), change the outgoing VoIP called number to the extension of the cookies agent, say 202. The table would look like this:

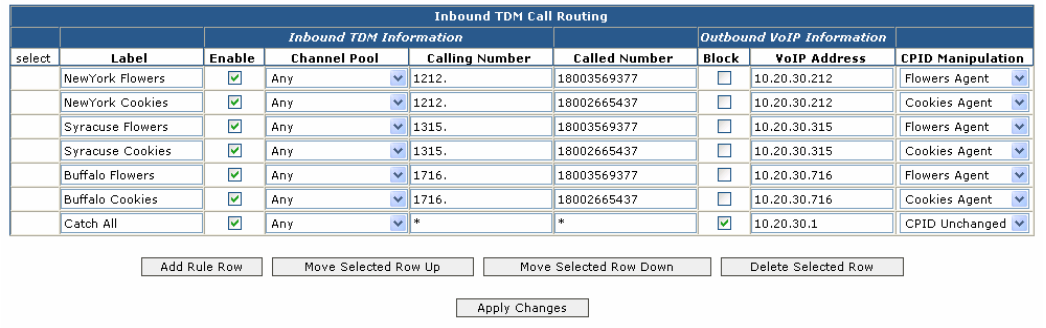
Figure 27. Setting Up the CPID Manipulation Rules



2. **Set up the TDM to VoIP routing rules.** Calls from the 212 area code route to the New York Distributor. Calls from the 716 area code route to the Buffalo Distributor. Calls from the 315 area code route to the Syracuse Distributor.

   Compare the first 4 digits of the incoming calling number to determine the VoIP address of the distributor. The table would look like this:

Figure 28. Setting Up the TDM to VoIP Routing Rules



Media Gateway Parsers 6

This section describes the Dialogic® Media Gateway in-band Type I (on-hook) and Type II (off-hook) integration parsers for analog and T1 CAS integrations, and the display parsers for digital integrations. These parsers allow the user to define the meaning of either the in-band/on-hook integration strings or display strings received from the telephony network. For this discussion, the term parser will be used when referencing all three variants listed above. Various options for entering the configuration data, parser syntax rules, and several examples are presented.

Note: The information in this section applies to all models of the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) models using the CAS protocol.

Information about the Media Gateway parsers is included in the following sections:

- Configuration Options 163
- Parsing Configuration Syntax 164

The parser allows the user to enter rules that define the meaning of the strings that can be received from the telephony network. The Media Gateway will use these rules to parse the strings and extract the source party information, destination party information, call reason (direct, busy, ring-no-answer, etc.), and call origin (internal or external). Note that the default rules supplied for the display parsing on digital units handle most scenarios, but the analog CPID parsing default rules are only included to provide examples for the user.

The user configurable rules (collectively known as the configuration data) contain expressions that represent the various types of integration strings available on the telephony switch interface. The configuration data contains rules and expressions that specify the location of the source, destination, reason, and origin information in the integration or display string.

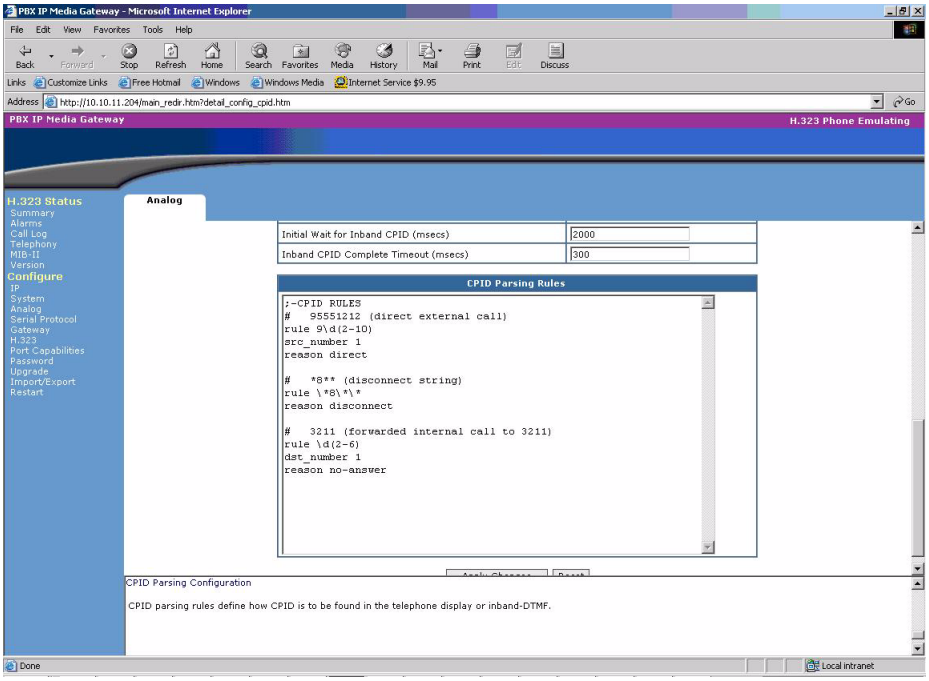
6.1 Configuration Options

The user can access the parsing configuration data in several different ways:

- Media Gateway configuration file (*.ini) which allows the user to use the import/export Web page to upload new configuration data or to download existing configuration data for review.
- Text file (must have an .adt file extension) containing only the configuration data.
- The cpid.htm hidden Web page.
- For type I and II analog parsing rules only, the analog page on the Web interface.

Figure 29 shows the default analog parsing configuration (used for either type I or II CPID) on the analog Web page. Figure 30 shows this same data as it appears in the *.ini file. Figure 31 shows a sample, downloadable text based *.adt file. Figure 32 shows the generic CPID configuration page available to all emulating units with the example showing the default Mitel digital display parsing rules.

Figure 29. Default Analog CPID Configuration Data on Analog Web Page



6.2 Parsing Configuration Syntax

The configuration data syntax is used to describe the type I or II CPID (analog or T1 CAS) or display (digital) strings that may be received from the telephone network. The type of information that can be received using analog DTMF integrations is a subset of that which can be gleaned from digital displays. For this reason, the rule syntax used for analog parsing is a much simpler subset of the digital parsing syntax.

The overall rule syntax is a subset of PERL. The syntax supports the extraction of the following information:

- Source Party Information (may include ANI information) - for analog DTMF integrations, this can only be numeric, whereas for digital integrations, the information can be a number and/or a name.
- Destination Party Information (may include DNIS information) - for analog DTMF integrations, this can only be numeric, whereas for digital integrations, the information can be a number and/or a name.

Figure 30. Default Analog CPID Configuration in the .ini File

```
;********************  PBX IP Media Gateway  *****************
;Version Information:
;MAC:00-a0-e6-06-05-07
;IP:10.0.1.41
;DSP Firmware: |9.1w/Fax|FRI MAY 02 17:59:51 2003|
;DSP Firmware (ROM): 9.1 w/Fax|FRI MAY 02 17:59:51 2003
;Main Board Boot (ROM): |4.2||WED APR 02 17:51:55 2003
;Gateway Application (ROM): |lab||FRI JUL 02 14:18:13 2004
;Main Board CPLD: Platform 3 Ver 1
;Adept Config: Default
;*********************************************************************
;Client IP Address: IP Adress in dotted decimal notation
ipClientAddr = 10.0.1.41
  *
  *
  *
;Call Progress Cadence Time Deviation (ms):Number between 0 - 1000
cpToneTimeDeviation: Tone Definition1 Time1 = 0
cpToneTimeDeviation: Tone Definition1 Time2 = 0
cpToneTimeDeviation: Tone Definition1 Time3 = 0
  *
  *
  *
cpToneTimeDeviation: Tone Definition16 Time1 = 0
cpToneTimeDeviation: Tone Definition16 Time2 = 0
cpToneTimeDeviation: Tone Definition16 Time3 = 0

;-CPID RULES
# 95551212 (direct external call)
rule 9\d(2-10)
src_number 1
reason direct

# *8** (disconnect string)
rule \*8\*\*
reason disconnect

# 3211 (forwarded internal call to 3211)
rule \d(2-6)
dst_number 1
reason no-answer
```

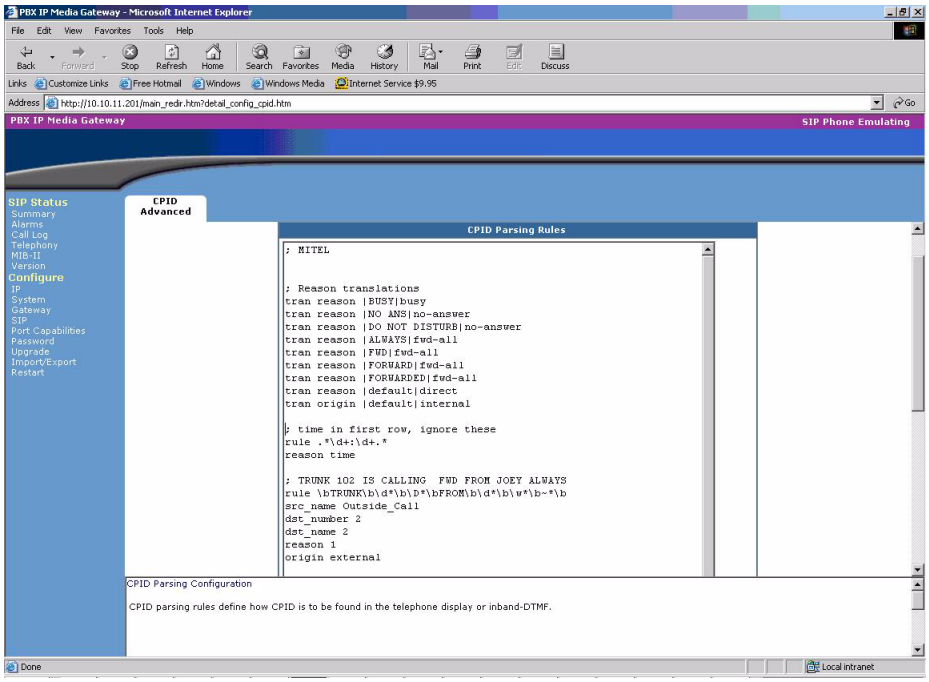
- Reason Code (no-answer, busy, etc.)
- Call Origin (internal, external)

The configuration data consists of call-class parsing rules. Call-class parsing rules are PERL-like expressions that define the incoming integration or text strings. The configuration data may contain multiple rules, each representing a different type of integration or display string in which the call party information exists in a different format (either textual representation or delimited fields).

Figure 31. Sample Analog Type II CPID Configuration Data in the .adt File

```
#### ATT System 25
rule #00#\d(1-5)##
src_number 1

rule #00#\d(1-5)#\d(1-5)#
src_number 1
dst_number 2
reason no-answer

rule #01#\d(1-3)##
reason trunk

rule #01###
reason opening

rule #01##
reason opening

rule #01#
reason opening

rule #02#\d(1-5)#\d(1-5)#
src_number 1
dst_number 2
reason no-answer

rule #03##\d(1-5)#
dst_number 1
reason no-answer

rule #04##\d(1-3)#
reason trunk

rule #05#\d(0-10)#\d(0-10)
reason disconnect
```

Figure 32. Default Mitel Digital CPID Configuration Data (cpid.htm)



The parser attempts to match an input integration or display string to a rule defined by the configuration data. If a match is made, the parser uses the call-party, reason code, and origin specifiers of the matching rule. In this manner, the parser can extract the call party information from the correct locations of the integration or display strings.

6.2.1 Display Translation Descriptors

Translation descriptors define translations between telephone switch-specific display tokens and strings that the application uses. Translation descriptors are global translations that govern all display parsing rules. The following is a reason code example.

```
trans reason |FWD|fwd-all
```

The above provides a translation of the switch-specific reason code "FWD" to the application string "fwd-all". If the token "FWD" is found in the reason-code section of a display, the reason code presented to the application will be "fwd-all". This provides a means of normalizing different switch-specific reason codes into strings that can be recognized by the application. The following is an origin example.

```
trans origin |default|internal
```

If the switch-specific code is 'default', then the application string of the translation is set as the default translation result if no switch-specific string is found. In the above case, the default call origin for all displays is 'internal'.

6.2.2 Call Class Rules

On a telephone network, there can be several different string formats for call information. Integration or display strings which use the same format are said to be in the same "Call Class". That is, two integration or display strings can be parsed using the same rules if they are in the same call class. If an integration or display string cannot be parsed using the same rules, then a new call class must be declared.

Regular Expressions

Call classes will be recognized using user-defined regular expressions. The parser uses standard regular expression "metacharacters" (special characters which are used to describe sequences of regular characters) in addition to some specific metacharacters.

Table 5. Parser Regular Expressions

Characters and Meta-Characters	Purpose	Analog Parsing Applicable
.	Any single character.	Yes
\	Character to right is literal. (quoted character)	Yes
?	0 or 1 of preceding character.	No
*	0 or more of preceding character.	Yes
+	1 or more of preceding character.	No
~	A reason code or call origin character.	No
\d	Digit character. (call party number)	Yes
\D	Non-digit character.	No
\w	Word character. (call party name) Must start with alpha, then can be alpha, digit, -, comma, space, ., or _.	No
\s	Whitespace character. Space, newline, or tab.	No
\b	Word boundary. Whitespace, punctuation, beginning or end of text. Used to specify that strings are bounded at start or end of string. Not explicitly searched for. Checked on reasons, digits, and literal strings.	No
\nnn	Octal control character.	No
(min,max)	Braces. Follows other chars. If non-null, length must be within min/max.	Yes

Rule Syntax

A call-class rule starts with the tag "rule". All characters following the tag define the rule. Following the rule are specifiers that define the location in the integration or display string of the call party information.

For example, to describe the integration string:

```
[A1500*300#]
```

the rule might read:

```
rule A1\d(0-10)\*\d(0-10)*#
dst_number      1
src_number      2
reason          no-answer
```

This means that when the regular expression is satisfied, the first digits string is the destination number, the second digit string is the source number, and the reason code is 'no-answer'.

In order to describe the display:

```
[a= JOE 123 to BILL 456 b]
```

the rule might read:

```
rule .= \w*\s\d* to \w*\s\d*\s*~*
src_number      1
src_name        1
dst_number      2
dst_name        2
reason          1
origin          internal
```

This means that when the regular expression is satisfied, the first digits string is the source number, the second digits string is the destination number, the first word string is the source name, the second word string is the destination name, and the reason code is at the end. The origin of the call is "internal".

Rule Order

The parser attempts to match an input integration or display string with a rule contained in the configuration data. The rules are compared to the specified string from the top rule to the bottom rule. Because of this, rules that contain the most specific information should be listed in the configuration file first. In this way, exact matches can occur before a more generic 'catch-all' rule is reached.

If a rule does not match, the parser attempts to match the next rule in the list. If no rules match, then the integration parsing fails. For this reason, the last rule in the rule list should be a very generic rule that provides the best possibility of extracting the desired information from the integration string.

Reason Tokens

The reason token specified in a rule is translated by the DMG1000 into corresponding IP call information. The following reason token strings specified in a rule are recognized by the DMG1000:

Table 6. Parser Reason Codes

Reason String	Used by Media Gateway
no-answer	Call is tagged as a forwarded on no-answer call.
busy	Call is tagged as a forwarded on busy call.
direct	Call is tagged as a direct call.
fwd-all	Call is tagged as a forwarded all call.
disconnect	Call is disconnected.

Any other call reason token specified in a rule will cause the Media Gateway to ignore the received string (if the received string matches the rule).

Data Security 7

Information about data security and how it is supported by the Dialogic® Media Gateway is described in the following sections:

- Data Security Overview 171
- Secure HTTP 171
- SIP Call Control Security using TLS 173
- Secure Voice Data 176
- Installing Certificate Using Internet Explorer 178

7.1 Data Security Overview

Data security on the Media Gateway includes the use of various secure protocols when transmitting and receiving data. The Media Gateway secures three types of data:

- HTTP - The data transmitted between the Media Gateway and a Web browser. To secure HTTP, the Media Gateway uses the Secure HTTP (HTTPS) protocol.
- Call Control - The data used to setup and tear down a call. To secure Call Control, the Media Gateway uses Transport Layer Security (TLS) on top of SIP
- Voice - The actual conversation once a call is connected. To secure voice, the Media Gateway uses Secure RTP (SRTP).

Note: The Media Gateway does not support security for the H.323 protocol. If H.323 protocol is selected, TLS and SRTP are not supported and call control and voice data can't be secured.

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

The HTTPS and TLS protocols require digital identity certificates (e.g. public key certificates). Therefore, certificate management is also covered in this section.

7.2 Secure HTTP

HTTP data is transmitted as messages between the Media Gateway and a Web browser. These messages travel on the network as clear text and can be "listened" to by anyone. Even though the HTTP interface has access security (via a password), privacy is not secure.

As an example, if a message containing a request to change a password were captured by a hacker or third party, the hacker or third party could log on to the Media Gateway and change the configuration. HTTPS safeguards HTTP data by encryption and authentication. With HTTPS, messages are no longer transmitted as clear text and are not readily readable.

HTTPS requires two actions by the user:

- Both the Media Gateway and the PC on which the Web browser used to connect to the Media Gateway via HTTPS is running must be configured with the proper certificate.
- When accessing the Media Gateway, use https:// instead of the non-secure http:// followed by the Media Gateway's URL.

This section includes the following information about HTTP security:

- HTTPS Certificate Configuration
- HTTPS Example

7.2.1 HTTPS Certificate Configuration

An HTTPS certificate can be either self-signed or certificate authority (CA) signed. A self-signed certificate can be generated by the Media Gateway. CA signed certificates must be requested by the Media Gateway and then signed by a CA.

When using a self-signed certificate:

- The Media Gateway generates a self-signed public key certificate.
- This certificate is then exported and downloaded from the Media Gateway to a PC via HTTP (or HTTPS if already active).
- The certificate is then configured into the Windows® PC running the HTTPS Web browser used to connect to the Media Gateway.
- From this PC, the user logs on to the Media Gateway using the https://[URL].
- HTTPS is then automatically used when accessing all subsequent Web pages.

When using a CA signed certificate:

- The Media Gateway generates a certificate signature request (CSR).
- The CSR is exported from the Media Gateway to a PC via HTTP (or HTTPS if already active).
- The CSR is used by the CA to create a signed certificate.
- The CA signed certificate is uploaded to the Media Gateway.
- The root certificate of the CA that signed the CSR is configured into the PC running the Web browser used to connect to the Media Gateway via HTTPS.
- The user logs into the Media Gateway by going to https://[URL]
- HTTPS is automatically used when accessing all the subsequent Web pages

The choice of either self-signed or CA-signed certificates depends on the system administration and the desired level of trust within the system. Self-signed certificates are generated by the Media Gateway and therefore do not cost any money - and may require less time to install. A self-signed certificate is simply downloaded from the gateway and installed on the PC running the Web browser used to connect to the Media Gateway via HTTPS.

However, when self-signed certificates are used, the PC/Web Browser must have a unique certificate installed for each Media Gateway with which it will communicate. This process could

get lengthy if the PC/Web Browser needs to communicate with a number of Media Gateway units. On the other hand, CA signed certificates require time and effort to install since the certificates must be signed by a CA. However, once you have the signed certificate, the CA root certificate can be used to communicate with multiple Media Gateway units.

7.2.2 HTTPS Example

An example of how HTTPS is used with a self-signed certificate is described below. In the example, the Media Gateway has an IP address of 172.16.3.10 and uses a self-signed certificate.

1. Start the Media Gateway.
2. Start Internet Explorer (or any Web browser that supports HTTPS).
3. In the Web browser Address box, enter http://172.16.3.10.
4. At the login screen, enter a User name and Password, and click OK to login to the Media Gateway.
5. Select the Security Web page > Certificate Management tab > Certificate Usage table > HTTPS parameter and check that **Self Signed** is the selected value.
6. Go to the HTTPS table and click on Generate button on the Self Signed row. The Self Signed Certificate Generation screen will appear.
7. Fill in the text boxes in the Value row of both the Certificate X509 Extensions and Certificate Subject tables and then click the Generate button. After a short time, the following message will appear: "Self signed Certificate was created". Click Continue to return to the Certificate Management Web page.
8. In the HTTPS table, click the Export button in the **Action** column of the **Self Signed** row to download the certificate from the Media Gateway to the PC.
9. Configure this certificate on the PC running the Web browser used to connect to the Media Gateway via HTTPS. See Section 7.5, “Installing Certificate Using Internet Explorer”, on page 178 for details.
10. In the Web browser Address box, enter https://172.16.3.10.
11. At the login screen, enter a User name and Password. Then click OK to login to the Media Gateway. HTTPS is now active.

7.3 SIP Call Control Security using TLS

This section includes the following information about SIP Call Control security using TLS:

- TLS Certificate Configuration
- TLS Feature Configuration
- TLS Examples

SIP is an application protocol used for VoIP call control. SIP messages are used for call setup and tear down. These messages contain information such as call-party information, call media type, whether it is a secure call, and if so, what encryption algorithm is used, etc. The SIP protocol can be carried by UDP, TCP, or TLS transports. Both UDP and TCP transport data in clear text. As a result, UDP and TCP can easily be monitored by third party hackers. TLS, on the other hand,

carries SIP data in a secure way by encrypting the data and authenticating the transport connections. Authentication guarantees that you are talking to the intended peer.

7.3.1 TLS Certificate Configuration

A TLS certificate can be self-signed or certificate authority (CA) signed. A self-signed certificate can be generated by the Media Gateway. CA signed certificates must be requested by the Media Gateway and signed by a CA.

When using a self-signed certificate:

- The Media Gateway generates a certificate which will be installed on VoIP devices that will communicate with the Media Gateway via TLS.

When using a CA signed certificate:

- The Media Gateway generates a certificate signature request (CSR) to a PC.
- The CSR is used by the CA to create a signed certificate.
- The root certificate of the CA that signed the CSR is uploaded to the Media Gateway along with the CA signed certificate.
- The root certificate of the CA that signed the CSR, as well as the signed certificate, are also configured into the VoIP devices that will communicate with the Media Gateway via TLS.

The choice of either self-signed or CA-signed certificates depends on the system-administration and the desired level of trust within the system. Self-signed certificates are generated by the Media Gateway and therefore, do not cost any money - and may take less time to install. A self-signed certificate is simply downloaded from the gateway and installed on VoIP devices that will communicate with the Media Gateway via TLS. However, when self-signed certificates are used the VoIP device must have a unique certificate installed for each Media Gateway with which it will communicate. This process could become lengthy if the VoIP device needs to communicate with a number of Media Gateway units. On the other hand, CA signed certificates require time and effort since the certificates must be signed by a CA. However, once you have the signed certificate, the CA root certificate can be used to communicate with multiple Media Gateway units.

7.3.2 TLS Feature Configuration

TLS has the following configurable features. The values of these configuration parameters can usually be left as default.

- SNTP Server IP Address - A server that the Media Gateway gets current time from to compare to the expiration date of a certificate. This is how the Media Gateway identifies an expired certificate when necessary. The expired certificates are identified by certificate date verification. This time-providing server is needed if a TLS certificate date is verified.
- TLS Transport Enabled - This parameter enables use of the TLS protocol and must be set to ‘Yes’.
- TLS Server Port - This is the IP port post number to listen to for TLS connection requests. Any number between 1024 and 65000 is valid. The default is 5061. If you wish to use a port

number other than the default, specify the number. The Media Gateway will then communicate this number to peers via URI.

- TLS Cipher List - The Cipher list is not a configurable parameter. The Media Gateway supports 6 ciphers in a list but cannot be changed. Valid OpenSSL ciphers can be found at: *http://www.openssl.org/docs/apps/ciphers.html*

  A default cipher list must be specified for TLS to work. The Media Gateway uses the default: ALL:!ADH:!LOW:!EXP:!MD5:@STRENGTH.
- SSL TLS Protocol - This parameter specifies the SSL record type to be used with the TLS connections and can be set to use SSLv3 and/or TLSv1.
- Mutual Authentication Required - Mutual Authentication Required. This parameter should be set to 'Yes' if the user wants the Media Gateway to authenticate the VoIP endpoint that it is communicating with when the VoIP endpoint initiates a SIP session. Otherwise, this parameter can be set to 'No'.
- TLS Inactivity Timer - This will determine when to close a TLS port. Any number between 10 to 60000 milliseconds is valid.
- SIPS URI Scheme Enabled - Selects the URI scheme, SIP or SIPS, that the Media Gateway will use for outgoing SIP call-requests. This may be limited by the capability of the other party that the Media Gateway communicates with. The Media Gateway accepts both SIP and SIPS URI schemes.
- Verify TLS Peer Certificate Date - If enabled, the peer certificate date is verified to detect if the peer certificate has expired. If so, the call request will be rejected. This indicates whether or not the certificate date is verified. Enable this feature if you want to detect expired certificates. Otherwise, keep it disabled. This feature will work correctly only if a SNTP server is available.
- Verify TLS Peer Certificate Trust - This indicates whether or not a certificate trust is verified. A certificate trust is the identity that signs the certificates. If the Media Gateway only accepts certificates signed by certain CA, then the Media Gateway compares the trust on a certificate to its trust list. If the trust is found in the list then the verification will pass. Enable this feature to increase security. However, if the other party that the Media Gateway communicates with is not capable of generating a trust, this feature must be disabled to avoid verification failure. You may also leave this feature disabled if encrypting data is sufficient.

7.3.3 TLS Examples

The following examples show how the Media Gateway should be configured for using TLS.

Example 1:

Assume the Media Gateway is talking to an IP phone that requires TLS and supports SIPS URI.

The Media Gateway can be configured as follows:

- SNTP Server IP Address: Leave blank
- TLS Inactivity Timer: Use default value
- TLS Server port: Use default value
- SIPS URI Scheme Enable: Yes
- Cipher List: Use default value

- Verify TLS Peer Certificate date: No
- Verify TLS Peer Certificate trust: No

Example 2:

Assume a company has five sites: one in Indiana, one in Illinois, one in California, one in New York, and one in Washington. Each of these sites uses a local telephone company. People at different sites frequently call each other. They have decided to use VoIP service with one Media Gateway at each site. They also use one media server. A CA signed certificate, which will expire in 1 year, is used. There is no convenient SNTP server. Each of the five Media Gateways can be configured as follows:

- SNTP Server IP Address: Leave blank
- TLS Inactivity Timer: Use default value
- TLS Server port: Use default value
- SIPS URI Scheme Enabled: Yes
- Cipher List: Use default value
- Verify TLS Peer Certificate date: No
- Verify TLS Peer Certificate trust: No

7.4 Secure Voice Data

This section includes the following information about secure voice data:

- Configuration
- Secure Voice Data Examples

Once a Voice over IP (VoIP) call is established, voice data is transported in the form of RTP packets. The voice data can be easily extracted from RTP packets and replayed using commercially available software. SRTP adds security by encrypting voice data and authenticating packets.

The two parties involved in a conversation must be "compatible" in the sense that each party understands the other party's cipher requirements and supports them. Configuration provides the following benefits:

- Support for more devices - For example, Media Gateway can talk to a device that supports either SHA1 32 bits authentication tag or SHA1 80 bits authentication tag.
- Turn security on or off completely. This allows the Media Gateway to talk to a device that doesn't support security at all.

7.4.1 Configuration

Secure RTP (SRTP) includes the following configuration parameters:

- SRTP Preference - Values for this parameter are SRTP_Only, RTP_Only, or SRTP_Preferred. A single setting applies to all channels. If SRTP_Only is specified, the gateway will only

request secure audio and will reject all requests for non-secure audio. If SRTP_Preferred is specified, the gateway will request both secure audio and non-secure audio, with a preference for secure audio, and the gateway will accept requests for both secure and non-secure audio. If RTP_Only is specified, the gateway will only request non-secure audio and will only accept requests for non-secure audio.

Note: SRTP_Preferred is not supported in Version 5.1 SU2 Software.

- Master Key Index (MKI) on Transmit Stream - With SRTP, audio data for a conversation is encrypted using a key that is called the session key. Each session key is derived from a master key communicated through SIP SDP. Multiple master keys may be used to add security. When multiple master keys are used, the master key index is used to identify a master key. When an audio data packet arrives, the master key index is specified in the packet so the packet receiver knows which key to use to decrypt the packet. Not all the devices support this feature.
- Key Derivation Enable - Key derivation refers to the process used to generate a session key from a master key. If the session key is generated once for each conversation, Key Derivation Enabled is set to false. Otherwise, if a session key is generated more than once from a master key, Key Derivation Enabled is set to true. Enabling Key Derivation provides more security, but not all devices support this feature.
- Key Derivation Rate (KDR) - This parameter is only usable if Key Derivation Enabled is set to Yes. KDR takes values of 16 to 24. When 0 is specified, the session key is only derived once, which is the same as when Key derivation Enabled set to No. However, if key derivation rate is to a value greater than 0, a new session key from the same master key is derived whenever the audio data packet index reaches the multiple of 2^{KDR}. For example, if KDR is 16, $2^{16} = 65536$. The session key will be derived whenever a packet index reaches 65536, 2*65536, 3*65536, etc…
- Cipher Mode - Cipher is the algorithm used to encrypt/decrypt a packet. The Media Gateway supports plain text or AES counter mode. When plain text is specified, no encryption/decryption is performed on the audio data and Authentication is skipped. By default, the Media Gateway supports the AES counter mode.

  Note: Plain_Text is not supported in Version 5.1 SU2 Software.
- Authentication Type - The Media Gateway supports no authentication or SHA1. It is recommended that you use authentication whenever the cipher is non-null as it provides more security.
- Authentication tag length. When SHA1 is used, the tag can be 32 bits or 80 bits. By default, the Media Gateway supports SHA1 80 bits.

7.4.2 Secure Voice Data Examples

The following examples show how the Media Gateway should be configured for SRTP.

Example1:

In this example, the Media Gateway is talking to an IP phone that requires security and supports the following:

- Cipher: AES Counter Mode
- Authentication: SHA1 32 bits or 80 bits

- MKI: Not supported
- KDR: Not supported

The Media Gateway should be configured as follows:

- SRTP Preference: SRTP Only
- Cipher Mode: AES Counter Mode
- Authentication Type: SHA1
- Authentication Tag Length: SHA1 32 bits or 80 bits
- Master Key Index (MKI) on Transmit Stream: No
- Key Derivation Enable: No
- Key Derivation Rate (KDR): Not applicable

Example 2:

In this example, the Media Gateway is talking to an IP phone that does not support security:

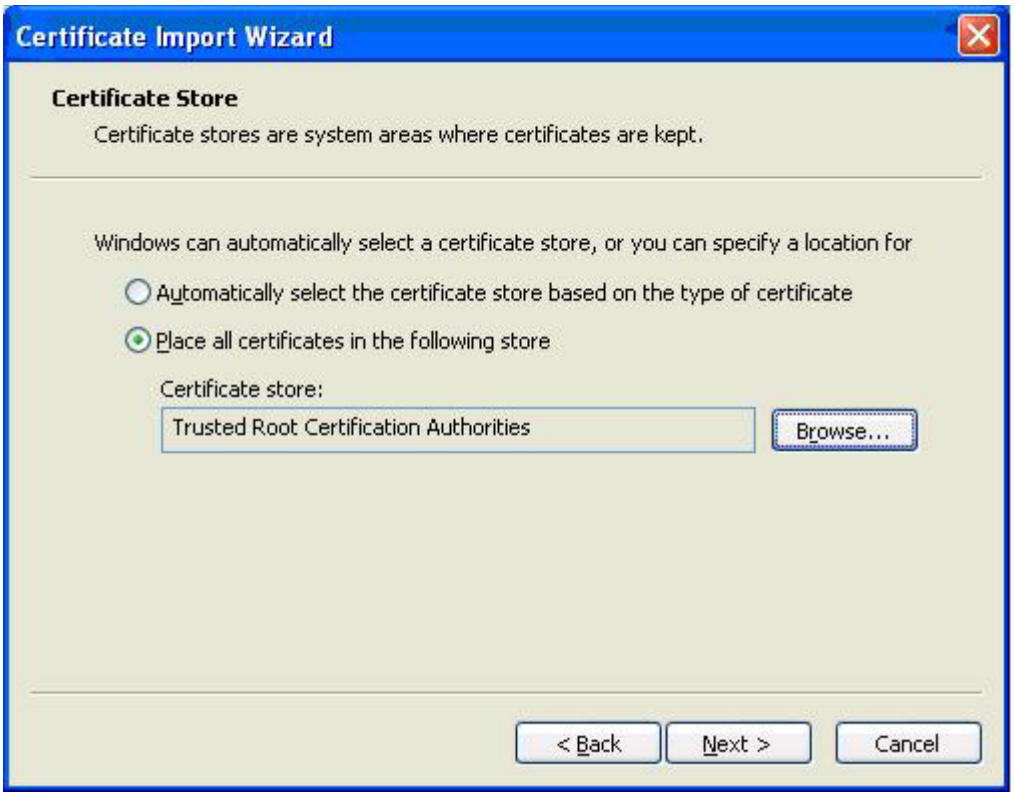
- SRTP Preference: RTP Only
- Master Key Index (MKI) on Transmit Stream: Not applicable
- Key Derivation Enable: Not applicable
- Key Derivation Rate (KDR): Not applicable
- Anti-replay window size hint: Not Applicable
- Cipher Mode: Not applicable
- Authentication Type: Not applicable
- Authentication Tag Length: Not applicable

7.5 Installing Certificate Using Internet Explorer

This section describes the procedure for installing a certificate using Internet Explorer. Perform the following steps:

1. Obtain a copy of the certificate that has already been installed or generated on the Media Gateway.
2. Launch Internet Explorer.
3. Select **Tools > Internet Options…** from the Internet Explorer menu.
4. Select the **Content** tab.
5. Click the **Certificates…** button.
6. Select the **Trusted Root Certificate Authorities** tab.
7. Click the **Import…** button.
8. Click **Next**.
9. Browse to the .cer file.
10. Click **Next**. You should see the following screen:

Figure 33. Storing Self-Signed Certificate by Certificate Import Wizard


Certificate Import Wizard
Certificate Store
Certificate stores are system areas where certificates are kept.
Windows can automatically select a certificate store, or you can specify a location for
Automatically select the certificate store based on the type of certificate
Place all certificates in the following store
Certificate store:
Trusted Root Certification Authorities
Browse...
< Back
Next >
Cancel

11. If the Certificate store is not **Trusted Root Certification Authorities**, use the **Browse** button to locate it.
12. Click **Next**.
13. Click **Finish**.
14. Click **Yes**.
15. Click the **OK** button.
16. Close the windows and Internet Explorer.

Unit Status 8

Status information about the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) units can be obtained through the Web interface.

The DMG1000 Status and DMG2000 Status Web pages allow an administrator to view the number of calls processed by the unit as well as a log of all calls (with call party information). The Media Gateway supports SNMP Version 1. The Media Gateway-specific SNMP MIB can be downloaded from the unit using the Web interface. The MIB file *dmg.mib* can be downloaded from the Media Gateway Status MIB-II Web sub-page.

Types of information available from the Status menu include:

- Summary Information 181
- Alarm Information 182
- Call Log Status Information 182
- Telephony Status Information 183
- MIB-II Status Information 183
- Version Information 192
- Diagnostics Information 193

8.1 Summary Information

Selecting Summary from the Status menu displays the following information:

- **Device** Summary Information:
  - The Media Gateway MAC address
  - The Media Gateway IP address
  - Uptime of the unit
  - Device Status
  - Ethernet connection losses
  - Telephony carrier losses
  - GateKeeper ID (if applicable)
- **Calls** Summary Information:
  - Current Calls
  - Inbound IP Calls
  - IP-to-PBX Calls
  - Inbound PBX Calls
  - PBX-to-IP Calls
- **Serial Protocol** Status (Enabled or Disabled)

8.2 Alarm Information

Selecting Alarm from the Status menu displays the following types of alarm information:

- Time- Timestamp of alarm entry.
- ID - Identifier of alarm entry.
- Severity - Severity Level of alarm entry.
- Description - Text Description of alarm entry.
- Port - Port Number of alarm entry.
- Text Message - Text message of alarm entry.

Clicking on the **Clear Alarms** button will clear all alarm information.

8.3 Call Log Status Information

Selecting Call Log from the Status menu displays information from the Call Log table, including:

- **ID** - Call Record ID number.
- **Start Time** - Displays the starting time of the call. This is the time at which the inbound call rang at the Media Gateway.
- **End Time** - Displays the end time of the call.
- **Source** - A call originates either from the PBX network or the IP network. This field specifies whether the call originated from the PBX network (From Switch Network) or from the IP network (From Packet Network).
- **End Reason** - Displays the reason for ending the call.
- **Inbound Info** - Displays the call party information of the inbound call. If the call is **From Switch Network**, this field contains the call party information of the inbound PBX call. In this case, the format of this field is:

  <PBX port number>:<calling party number>,<calling party name> -> <called party number>, <called party name>

  If the call is **From VoIP Network**, this field contains the call party information of the inbound IP call. In this case, the format of this field is:

  <calling party number>,<calling party name>, <calling party IP> -> <called party number>, <called party name>, <called party IP>
- **Outbound Info** - Displays the call party information of the outbound call. If the call is **From Switch Network**, this field contains the call information used to dial the outbound call to the IP network. In this case, the format of this field is:

  <alias or IP address dialed>

  If the call is **From VoIP Network**, this field contains the call information used to dial the outbound call to the PBX network. In this case, the format of this field is:

  <PBX port number>:<number dialed>

Clicking on the **Clear Log** button will clear all call information from the Call Log table.

8.4 Telephony Status Information

Selecting Telephony from the Status menu displays information about the state of each PBX port that connects to the Media Gateway. For each port, the following status information is provided:

- State (DMG1000):
  - In Service
  - No Link
  - Red Alarm
  - Yellow Alarm
  - Pend D-Chan
  - n/a
- State (DMG2000):
  - In Service
  - No Link
  - Red Alarm
  - Yellow Alarm
  - Pend D-Chan
  - n/a

Note: If the T1/E1 Signaling Mode parameter is set to ISDN, the Port # refers to the T1 or E1 span number.

8.5 MIB-II Status Information

Selecting MIB-II from the Status menu provides a number of Web subpages that include MIB-II information as documented in RFC 1213. The Web subpages include:

- System: Provides general Simple Network Management Protocol (SNMP) agent information including: description, object ID, up time, contact information, node name, node location, and the type of services offered.

  From the System Web subpage, you can download the Media Gateway MIB file *dmg.mib*.
- Interfaces: Provides Ethernet interface descriptions including: type, speed, physical address, and statistic counters.
- AT: Lists Address Translation tables (physical-to-network address mappings).

  Note: AT is not supported in Version 5.1 SU2 Software.
- IP: Lists counters and address tables for the Internet Protocol (IP) layer.
- ICMP: Provides Internet Control Message Protocol (ICMP) packet and error counter information.
- TCP: Provides information about the Transmission Control Protocol (TCP) counters and connection table.
- UDP: Provides information about the User Datagram Protocol (UDP) counters and local listener table.

- SNMP: Records error statistics for SNMP protocol datagrams.

8.5.1 SNMP Agent

The gateway SNMP agent is v1/v2c compatible. The agent provides read-only access to the gateway's configuration and status data. The gateway does not support SNMP write commands since SNMP v2 does not provide secure communications. You can configure the gateway through a Web browser using HTTP (or HTTPS when secure communication is required).

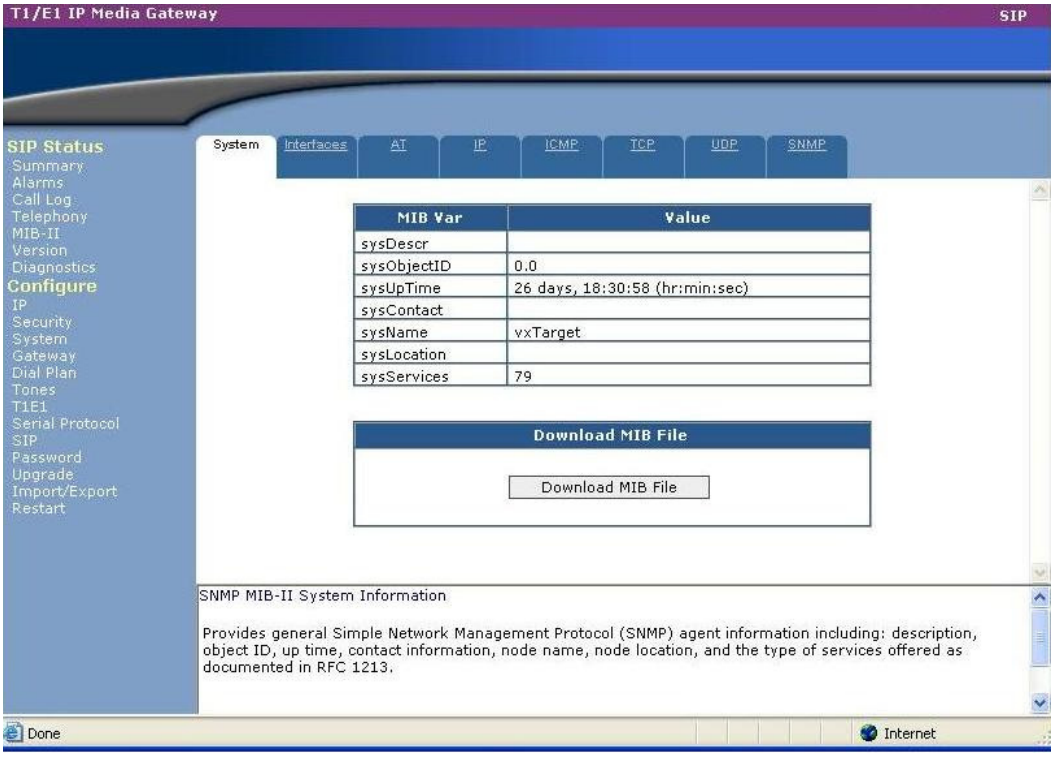
The SNMP agent provides GET, NEXT, BULK GET and V1 trap PDU's. The agent supports RFC 1213 MIB-II. The gateway also supplies a MIB file suitable for importing into your MIB browser.

8.5.1.1 Downloading the Media Gateway MIB File

Procedure to download Media Gateway MIB file to your computer:

From a Web browser you navigate to the gateway's MIB-II status page (Figure 34). Click on the download button and a save/cancel dialog opens. Click on save and the MIB file *dmg.mib* can be saved on your computer.

Figure 34. Media Gateway MIB File Download Web Page



8.5.2 Supported MIBs

8.5.2.1 RFC 1213 MIB-II

The gateway supports RFC 1213 MIB-II titled "Management Information Base for Network Management of TCP/IP-based Internets: MIB-II".

The RFC is available on numerous Web sites. A few are:

- *http://www.faqs.org/rfcs/rfc1213.html*
- *http://tools.ietf.org/html/rfc1213*
- *http://www.rfc-archive.org/getrfc.php?rfc=1213*

The Defense Data Network Network Information Center (DDN NIC) is the primary repository for RFCs. Hard copies can be obtained from them.

8.5.2.2 Gateway Specific MIB

Table 7 shows the gateway specific MIB where <p> = 1.3.6.1.4.1.343.2.121.1.

Table 7. Gateway Specific MIB

OID	Name	Position	Type
<p>.1	version	Node	
<p>.1.1	.versionTable	Node	
<p>.1.1.1	..versionTableEntry	Node	
<p>.1.1.1.1	…versionIdx	Leaf	Integer
<p>.1.1.1.2	…versionDesc	Leaf	Octet String
<p>.1.1.1.3	…versionVal	Leaf	Octet String
<p>.2	alarm	Node	
<p>.2.1	.alarmTable	Node	
<p>.2.1.1	..alarmEntry	Node	
<p>.2.1.1.1	…alarmIdx	Leaf	Integer
<p>.2.1.1.2	…alarmed	Leaf	Integer
<p>.2.1.1.3	…alarmSeverity	Leaf	Integer
<p>.2.1.1.4	…alarmDesc	Leaf	Octet String
<p>.2.1.1.5	…alarmDevice	Leaf	Integer
<p>.2.1.1.6	…alarmText	Leaf	Octet String
<p>.2.1.1.7	…alarmTime	Leaf	TimeTicks

Table 7. Gateway Specific MIB

OID	Name	Position	Type
<p>.3	status	Node	
<p>.3.1	.numCurrentCalls	Leaf	Integer
<p>.3.2	.numEthernetConnectionLosses	Leaf	Integer
<p>.3.3	.numInboundIpCalls	Leaf	Integer
<p>.3.4	.numInboundPbxCalls	Leaf	Integer
<p>.3.5	.numIpToPbxCalls	Leaf	Integer
<p>.3.6	.numPbxToIpCalls	Leaf	Integer
<p>.3.7	.numPeerConnectionLosses	Leaf	Integer
<p>.3.8	.numTelephonyCarrierLosses	Leaf	Integer
<p>.3.9	.statusSerialMasterClientTable	Node	
<p>.3.9.1	..statusSerialMasterClientTableEntry	Node	
<p>.3.9.1.1	…statusSerialMasterClientIdx	Leaf	Integer
<p>.3.9.1.2	…statusSerialMasterClientIp	Leaf	IpAddress
<p>.3.9.1.3	…statusSerialMasterClientLtns	Leaf	Octet String
<p>.3.10	.statusSerialSlaveConnection	Leaf	Octet String
<p>.3.11	.statusSerialSlaveMasterIp	Leaf	IpAddress
<p>.3.12	.statusDevice	Leaf	Octet String
<p>.3.13	.statusFreeMemBytes	Leaf	Integer
<p>.3.14	.statusGateKeeper	Leaf	Octet String
<p>.4	config	Node	
<p>.4.1	.configTable	Node	
<p>.4.1.1	..configTableEntry	Node	
<p>.4.1.1.1	…configIdx	Leaf	Integer
<p>.4.1.1.2	…configToken	Leaf	Octet String
<p>.4.1.1.3	…configInst	Leaf	Integer
<p>.4.1.1.4	…configSubInst	Leaf	Integer
<p>.4.1.1.5	…configVal	Leaf	Octet String
<p>.5	callLog	Node	
<p>.5.1	.callLogTable	Node	
<p>.5.1.1	..callLogEntry	Node	
<p>.5.1.1.1	…callLogIdx	Leaf	Integer
<p>.5.1.1.2	…callLogDirection	Leaf	Octet String
<p>.5.1.1.3	…callLogStartTime	Leaf	TimeTicks

Table 7. Gateway Specific MIB

OID	Name	Position	Type
<p>.5.1.1.4	…callLogEndTime	Leaf	TimeTicks
<p>.5.1.1.5	…callLogEndReason	Leaf	Octet String
<p>.5.1.1.6	…callLogInboundInfo	Leaf	Octet String
<p>.5.1.1.7	…callLogOutboundInfo	Leaf	Octet String

8.5.3 SNMP Alarms

Gateway alarms are reported as SNMP traps. They can also generate an e-mail as described in Section 8.5.3.1, "Alarm E-Mails", on page 189.

Table 8 maps an alarm severity index to a severity level. Table 9 shows the alarm information.

Table 8. Severity Index to Severity Level

Severity Index	Severity Level
1	Error
2	Warning
3	Information

Table 9. Alarm Information

Alarm ID			
Decimal	Hex	Severity	Description
0	0x0000	1	Gateway Error
1	0x0001	1	Gateway Software Fault
2	0x0002	1	Gateway Memory Unavailable
3	0x0003	1	Gateway Load Error
4	0x0004	1	Gateway Device Open Failure
5	0x0005	1	Gateway ARP Failure
6	0x0006	1	Gateway SNTP Time Query Failed
7	0x0007	1	Gateway Certificate Error
64	0x0040	2	Gateway Warning
65	0x0041	2	Gateway Memory Low
66	0x0042	2	Gateway Configuration Failure
67	0x0043	2	Gateway Peer Connection Failure
68	0x0044	2	Gateway Peer Connection Lost
128	0x0080	3	Gateway Information

Table 9. Alarm Information

Alarm ID			
Decimal	**Hex**	**Severity**	**Description**
129	0x0081	3	Gateway Initialized
130	0x0082	3	Gateway Restarted
131	0x0083	3	Gateway Upgrade Failed
132	0x0084	3	Gateway Memory Normal
133	0x0085	3	Gateway Peer Connection Gained
134	0x0086	3	Gateway MIB-II Alarm
135	0x0087	3	Gateway SNTP Time Query Completed
SIP			
256	0x0100	1	SIP Interface Error
257	0x0101	1	SIP Registration Failure
258	0x0102	1	SIP Resources Unavailable
259	0x0103	1	SIP TLS Initialization Failure
260	0x0104	1	SIP TLS Certificate Verification Failure
320	0x0140	2	SIP Interface Warning
321	0x0141	2	SIP Switched to Proxy Backup
322	0x0142	2	SIP Resources Low
323	0x0143	2	SIP Invalid Codec
384	0x0180	3	SIP Interface Information
385	0x0181	3	SIP Switched to Primary Proxy
386	0x0182	3	SIP Resources Normal
H323			
512	0x0200	1	H.323 Interface Error
513	0x0201	1	H.323 Registration Failure
514	0x0202	1	H.323 Resources Unavailable
576	0x0240	2	H.323 Interface Warning
577	0x0241	2	H.323 Resources Low
578	0x0242	2	H.323 Master-Slave Determination Reject
579	0x0243	2	H.323 Terminal Capability Reject
580	0x0244	2	H.323 Invalid Codec
640	0x0280	3	H.323 Interface Normal
641	0x0281	3	H.323 Resources Normal
TDM			
768	0x0300	1	TDM Interface Error
769	0x0301	1	TDM Interface Load Failure

Table 9. Alarm Information

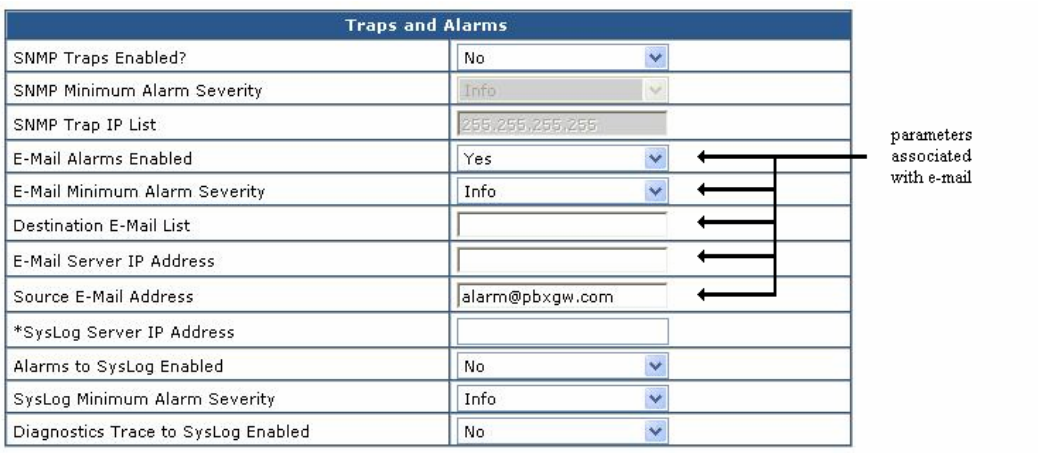
Alarm ID			
Decimal	**Hex**	**Severity**	**Description**
832	0x0340	2	TDM Interface Warning
833	0x0341	2	TDM Out of Service
834	0x0342	2	TDM Power Fault
835	0x0343	2	TDM Upgrade Required
836	0x0344	2	TDM Configuration Failed
837	0x0345	2	TDM Link Lost
838	0x0346	2	TDM MWI Failure
839	0x0347	2	TDM Invalid PCM Coding
896	0x0380	3	TDM Interface Information
897	0x0381	3	TDM Interface Initialized
898	0x0382	3	TDM Port Restarted
899	0x0383	3	TDM In Service
900	0x0384	3	TDM Link Gained
DSP			
1024	0x0400	1	DSP Interface Error
1025	0x0401	1	DSP T.38 Fax Resources Unavailable
1088	0x0440	2	DSP Interface Warning
1152	0x0480	3	DSP Interface Information
SRTP			
1280	0x0500	1	SRTP Rx Error
1281	0x0501	1	SRTP Tx Error

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

8.5.3.1 Alarm E-Mails

The gateway can be configured to generate an e-mail on an alarm. The necessary parameters are configured on the Gateway Advanced Web page using the Traps and Alarms section. Figure 35 shows the necessary parameters.

Figure 35. E-mail Parameters in Traps and Alarms Section of Gateway Advanced Web Page



E-Mail Alarms Enabled: Set to yes to generate alarm e-mails

E-Mail Minimum Alarm Severity: The gateway has three levels of alarms: Info, Warn and Error, where Info is the lowest level alarm and Error is the highest. This field specifies the lowest level that generates e-mails.

Destination E-Mail List: A list of e-mail addresses, separated by semi-colons, to which the alarm e-mails are sent.

E-Mail Server IP Address: The IP address of your network's e-mail server.

Source E-Mail Address: The e-mail address placed in the from field of the alarm e-mail. Default value is shown.

E-Mail Format

The e-mail's title specifies the alarm's severity, the gateway's IP address and a short description of the alarm. The e-mail's body starts with five standard lines followed by optional lines.

- **Line 1:** the alarm's severity (Info, Warn or Error)
- **Line 2:** the IP address of the gateway
- **Line 3:** the MAC address of the gateway
- **Line 4:** the alarm's identification number (in hexadecimal)
- **Line 5:** a short description of the alarm
- **Line 6:** the physical port number associated with the alarm (is omitted for alarms not associated with ports)
- **Remaining Lines:** detailed text associated with the alarm (present for a limited number of alarms that require additional information)

Sample E-Mails

Figure 36 shows the e-mail sent when the gateway powers up. This is an info alarm.

Figure 36. Sample Info Alarm E-Mail

Gateway Info Alarm on 172.16.3.150: Gateway Initialized
alarm@pbxgw.com [alarm@pbxgw.com]
Sent: Wednesday, January 31, 2007 5:09 PM
To: Mike Rupp
Gateway Info alarm on:
IP Address = 172.16.3.150
MAC Address = 00-0e-0c-ab-d2-f3
Alarm ID = 0x81
Alarm Description = Gateway Initialized

Figure 37 shows the e-mail sent when a port loses communication with the PBX. Notice that the associated physical port is shown. This is a warn alarm.

Figure 37. Sample Warn Alarm E-Mail

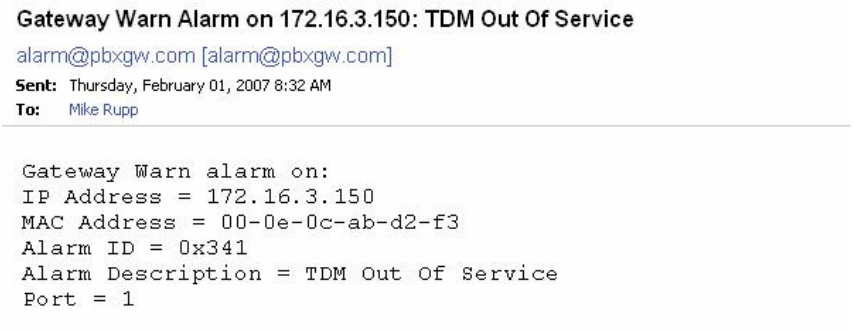
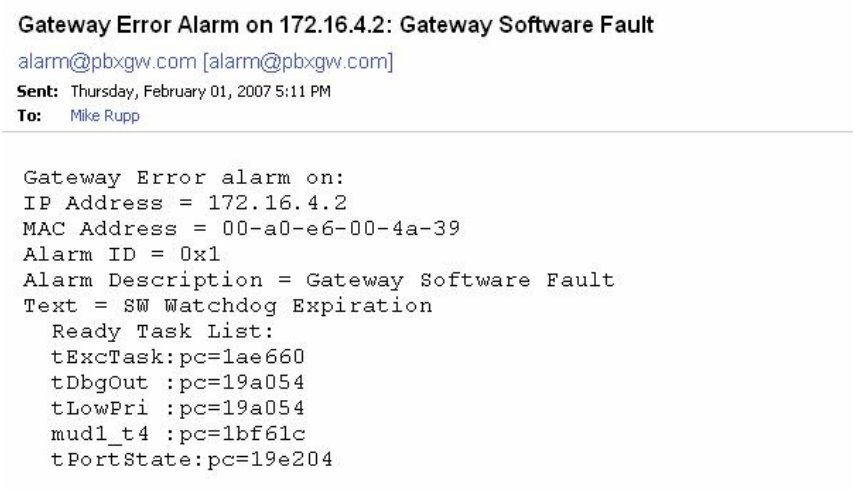

Gateway Warn Alarm on 172.16.3.150: TDM Out Of Service
alarm@pbxgw.com [alarm@pbxgw.com]
Sent: Thursday, February 01, 2007 8:32 AM
To: Mike Rupp
Gateway Warn alarm on:
IP Address = 172.16.3.150
MAC Address = 00-0e-0c-ab-d2-f3
Alarm ID = 0x341
Alarm Description = TDM Out Of Service
Port = 1

Figure 38 shows the e-mail sent for a software fault, more specifically a SW Watchdog Timer Expiration. Notice that it has a more detailed description. This is an error alarm.

Figure 38. Sample Error Alarm E-Mail



Gateway Error Alarm on 172.16.4.2: Gateway Software Fault

alarm@pbxgw.com [alarm@pbxgw.com]

Sent: Thursday, February 01, 2007 5:11 PM

To: Mike Rupp

```
Gateway Error alarm on:
IP Address = 172.16.4.2
MAC Address = 00-a0-e6-00-4a-39
Alarm ID = 0x1
Alarm Description = Gateway Software Fault
Text = SW Watchdog Expiration
  Ready Task List:
  tExcTask:pc=1ae660
  tDbgOut :pc=19a054
  tLowPri :pc=19a054
  mud1_t4 :pc=1bf61c
  tPortState:pc=19e204
```

8.6 Version Information

Selecting Version from the Status menu displays version information about the Media Gateway software and hardware.

Note: The firmware version information displayed will vary, depending on the Media Gateway model.

Version information includes the following:

- Gateway Application (ROM)
- Gateway Application
- Main Board Boot (ROM)
- DSP Firmware (ROM)
- DSP Firmware
- Telephony Interface Firmware (ROM)
- Telephony Interface Firmware
- Adept Config (ROM)
- Telephony Interface ID
- Port Flags
- Adept Config
- Telephony Interface Application
- Telephony Interface Boot
- Certificate Bundle (ROM)

8.7 Diagnostics Information

Selecting Diagnostics from the Status menu displays diagnostics information about the Media Gateway. Refer to Chapter 9, "Diagnostics" for detailed information about the Media Gateway diagnostic capabilities.

Diagnostics 9

This chapter describes how to perform diagnostics tasks on the Dialogic® 1000 Media Gateway (DMG1000) and Dialogic® 2000 Media Gateway (DMG2000) units and includes the following sections:

- VoIP Interface Test 195
- PBX/PSTN Interface Test 199
- PBX Self Verification Test 203
- Diagnostic Logging 213
- Communicating to the Terminal Interface. 222
- Trace Mechanism 223
- Diagnostic Commands. 235

9.1 VoIP Interface Test

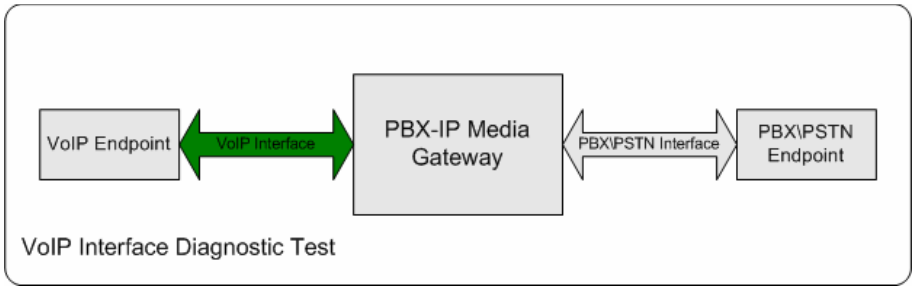
VoIP Interface Test is discussed in the following topics:

- VoIP Interface Test Overview
- VoIP Interface Test Operation

9.1.1 VoIP Interface Test Overview

The VoIP interface diagnostic tool is a Web-based user application that can be used to provide verification of VoIP compatibility or troubleshoot the VoIP interface of the Media Gateway.

Figure 39. VoIP Interface Diagnostic Test



9.1.1.1 Features

Currently the VoIP interface diagnostic provides the ability to:

- Originate a VoIP call to a specified endpoint address and verify endpoint response.
- Send a message waiting notification status update request and verify endpoint response.

9.1.1.2 Location

The VoIP interface diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.1.2 VoIP Interface Test Operation

To operate the VoIP interface diagnostic tool the user must first navigate to the *VoIP Interface* tab via the Media Gateway's user interface.

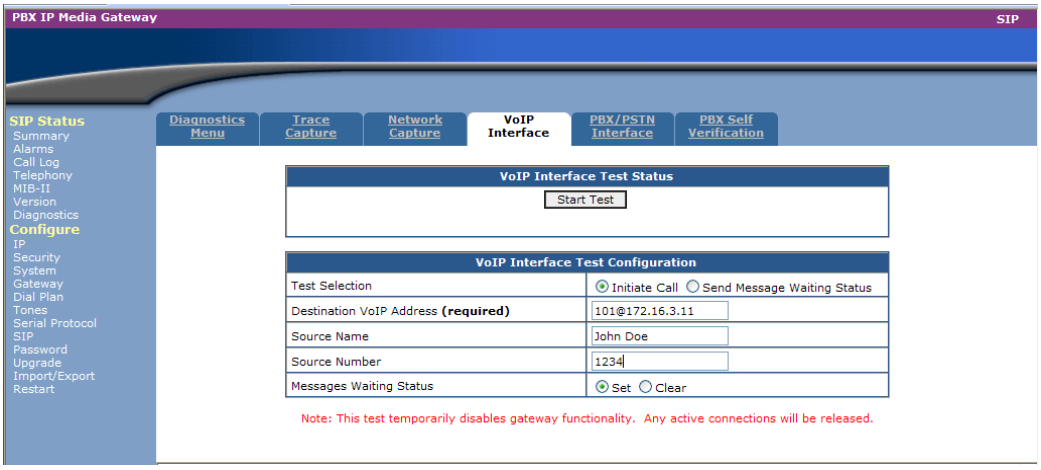
In the VoIP Interface Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field.
2. Then the user must fill in the **Destination VoIP Address** field of the VoIP endpoint that the diagnostic test will use when attempting to establish a connection.
3. Fill in any optional source party information (**Source Name** and **Source Number**) that will be delivered to the destination endpoint during the diagnostic test.
4. If the message waiting diagnostic test is selected, choose the **Messages Waiting Status** to be used in the message waiting notification request.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the VoIP interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 40. VoIP Interface Web Page



9.1.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Test Selection

Choose the type of diagnostic test to perform. Available options are *Initiate Call* and *Send Message Waiting Status*.

Destination VoIP Address

The **required** address of the VoIP endpoint that the diagnostic test will use when attempting to establish a connection. If this configuration is empty the diagnostic test will not be performed.

Source Name

The alphanumeric name representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Source Number

The number representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Messages Waiting Status

Choose the status value of the message waiting notification update. This field is only valid when the *Message Waiting* diagnostic test is selected.

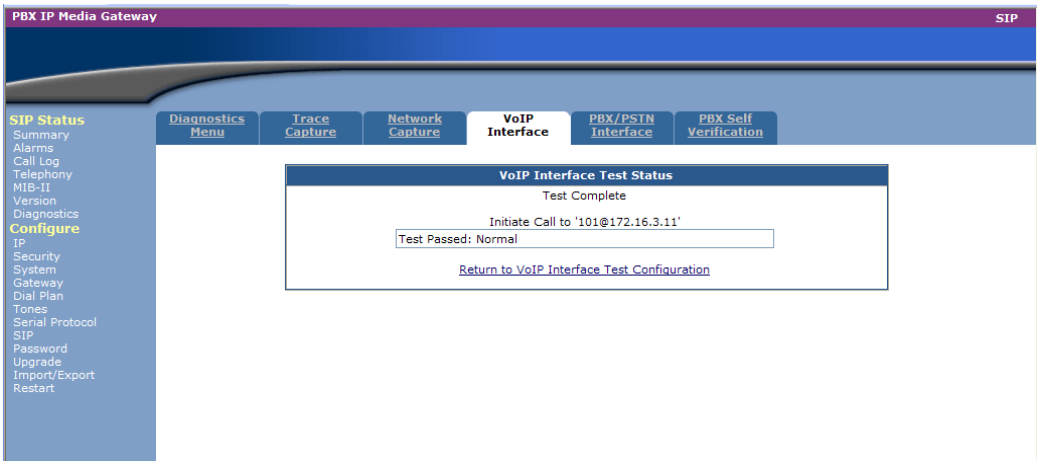
9.1.2.2 Test Status

When a diagnostic test is executed the test results are shown in the VoIP Interface Test Status table on the Web page.

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the VoIP Interface Test configuration Web page.

Figure 41. VoIP Interface Test Status Web Page



When the test is completed the results will show if the diagnostic test was able to establish a VoIP connection with the specified **Destination VoIP Address**.

Test Passed: Normal	VoIP connection was successful
Test Failed: *CallEndReason*	VoIP connection failed

The *CallEndReason* description is the error response received either from the specified destination VoIP endpoint or the VoIP interface of the gateway.

Some of the most common error responses and causes are:

Device Not Available	User at VoIP endpoint is not in-service
Transport Failed	VoIP address is invalid or no response received
User Not Found	User not found at VoIP endpoint
Ambiguous Request	VoIP address requires user information

If an error occurs some other suggestions are:

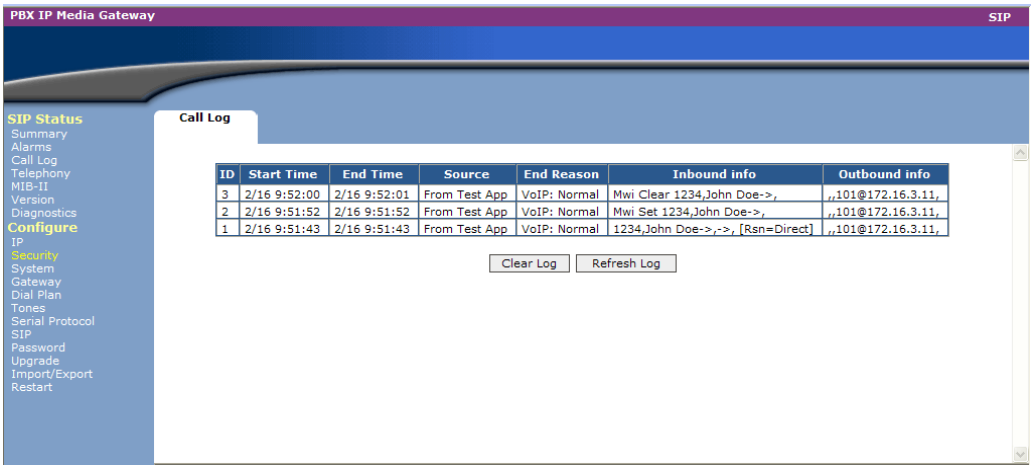
- Verify gateway VoIP interface is in-service and configured properly.
- Verify destination VoIP endpoint is valid and in-service.
- Run the test again with the **Trace Capture** and **Network Capture** diagnostic utilities started to log the detailed call flow of the VoIP interface diagnostic test.

9.1.2.3 Call Log

The test results from the VoIP interface diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 42. VoIP Interface Call Log Web Page



9.2 PBX/PSTN Interface Test

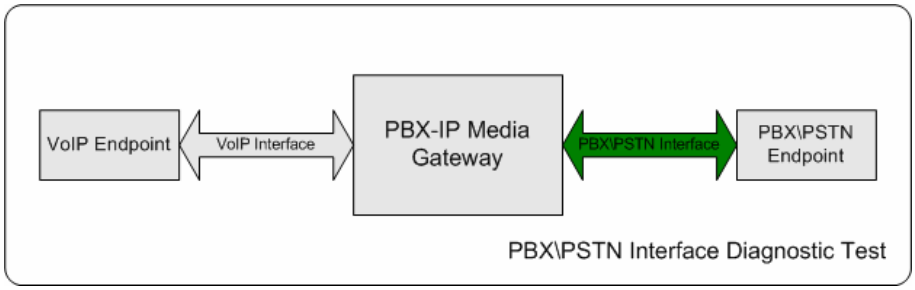
PBX/PSTN Interface Test is discussed in the following topics:

- PBX/PSTN Interface Test Overview
- PBX/PSTN Interface Test Operation

9.2.1 PBX/PSTN Interface Test Overview

The PBX/PSTN interface diagnostic tool is a Web-based user application that can be used to provide verification of PBX/PSTN compatibility or troubleshoot the PBX/PSTN interface of the Media Gateway.

Figure 43. PBX/PSTN Interface Diagnostic Test



9.2.1.1 Features

Currently the PBX/PSTN interface diagnostic provides the ability to:

- Originate a PBX/PSTN call to a specified endpoint address and verify endpoint response.
- Send a message waiting notification status update request and verify endpoint response.

9.2.1.2 Location

The PBX/PSTN interface diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.2.2 PBX/PSTN Interface Test Operation

To operate the PBX/PSTN interface diagnostic tool the user must first navigate to the *PBX/PSTN Interface* tab via the Media Gateway's user interface.

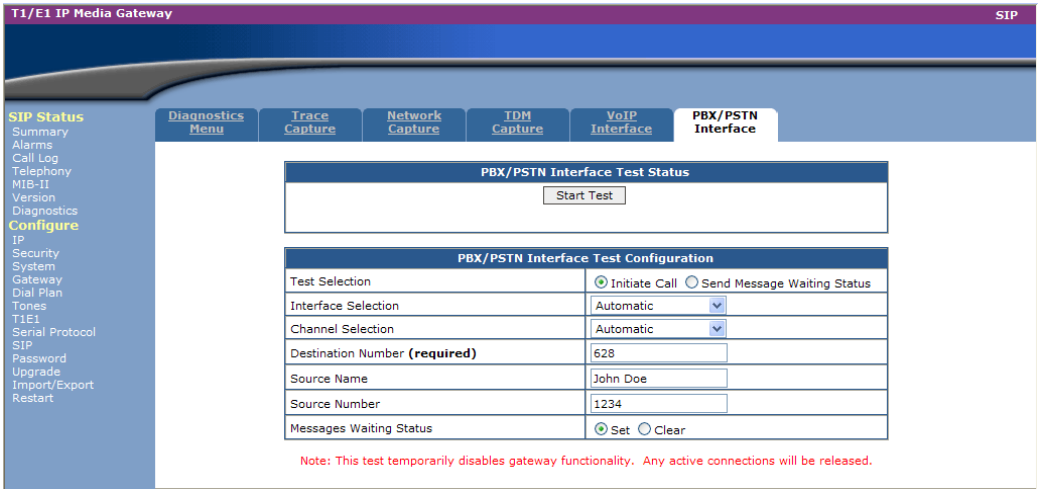
In the PBX/PSTN Interface Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field.
2. The user must fill in the **Destination Number** field of the PBX/PSTN endpoint that the diagnostic test will use when attempting to establish a connection.
3. Optionally, select the interface and channel index of the gateway to use in the diagnostic test by choosing the **Interface Selection** or **Channel Selection**. If *Automatic* is chosen the test will select the first interface and channel available.
4. Fill in any optional source party information (**Source Name** and **Source Number**) that will be delivered to the destination endpoint during the diagnostic test.
5. If the message waiting diagnostic test is selected, choose the **Messages Waiting Status** to be used in the message waiting notification request.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the PBX/PSTN interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 44. PBX/PSTN Interface Web Page



9.2.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Test Selection

Choose the type of diagnostic test to perform. Available options are *Initiate Call* and *Send Message Waiting Status*.

Interface Selection

Select the specific interface to use in the diagnostic test. If a specific interface is not chosen the test will select the first interface available.

Channel Selection

Select the specific interface to use in the diagnostic test. If a specific interface is not chosen the test will select the first interface available. This option is only available on PBX/PSTN interfaces that have more than one bearer channel.

Destination Number

The **required** number of the PBX/PSTN endpoint that the diagnostic test will use when attempting to establish a connection. If this configuration is empty the diagnostic test will not be performed.

Source Name

The alphanumeric name representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Source Number

The number representing the source party that is delivered to the destination endpoint during the diagnostic test. This is an optional field that is not required to perform the diagnostic test.

Messages Waiting Status

Choose the status value of the message waiting notification update. This field is only valid when the *Message Waiting* diagnostic test is selected.

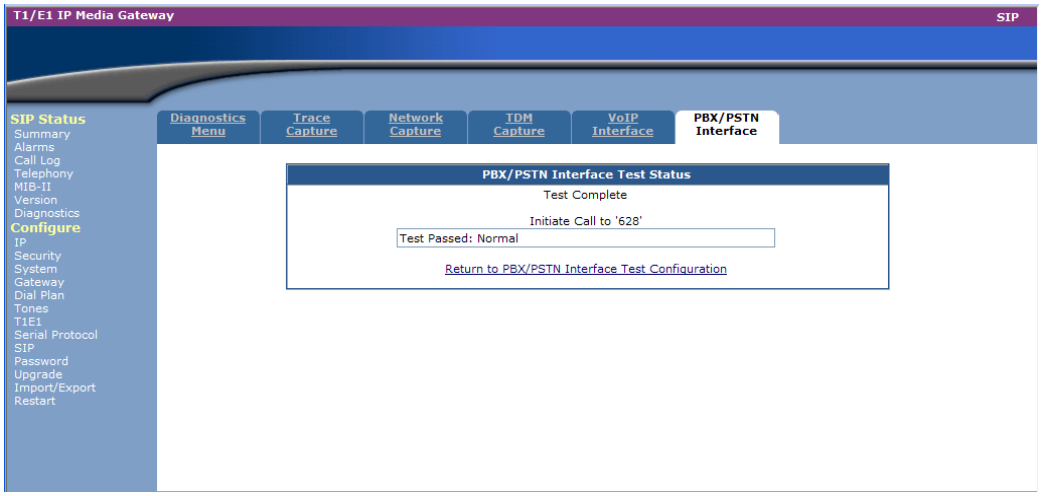
9.2.2.2 Test Status

When a diagnostic test is executed the test results are shown in the PBX/PSTN Interface Test Status table on the Web page.

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the PBX/PSTN Interface Test configuration Web page.

Figure 45. PBX/PSTN Interface Test Status Web Page



When the test is completed the results will show if the diagnostic test was able to establish a PBX/PSTN connection with the specified **Destination Number**.

Test Passed: Normal	PBX/PSTN connection was successful
Test Failed: *CallEndReason*	PBX/PSTN connection failed

The *CallEndReason* description is the error response received either from the specified destination PBX/PSTN endpoint or the PBX/PSTN interface of the gateway.

Some of the most common error responses and causes are:

Device Not Available	PBX/PSTN interface is not in-service
User Busy	Destination number is busy (in-use)
User Not Found	Destination number not found at PBX/PSTN endpoint

If an error occurs some other suggestions are:

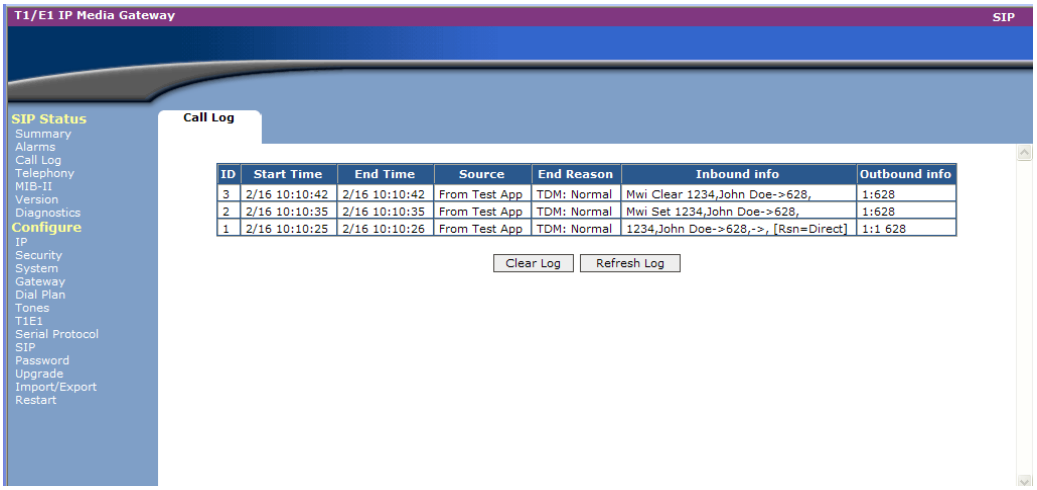
- Verify gateway PBX/PSTN interface is in-service and configured properly.
- Verify destination PBX/PSTN endpoint is valid and in-service.
- Run the test again with the **Trace Capture** and **Network Capture** diagnostic utilizes started to log the detailed call flow of the VoIP interface diagnostic test.

9.2.2.3 Call Log

The test results from the PBX/PSTN interface diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 46. PBX/PSTN Interface Call Log Web Page



9.3 PBX Self Verification Test

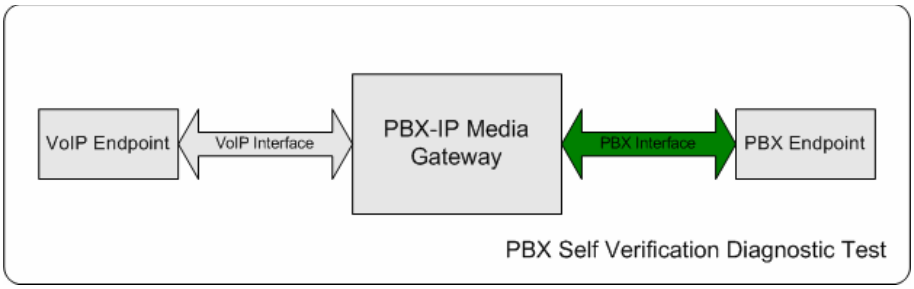
PBX Self Verification is discussed in the following topics:

- PBX Self Verification Test Overview
- PBX Self Verification Test Operation

9.3.1 PBX Self Verification Test Overview

The PBX self verification diagnostic tool is a Web page user application that can be used to provide verification of PBX compatibility and also to confirm the Media Gateway configuration values relating to the PBX interface are valid.

Figure 47. PBX Self Verification Diagnostic Test



9.3.1.1 Features

Currently the PBX self verification diagnostic provides the ability to verify the following operations:

Initiate Call

- Originate an outbound PBX call
- Receive outbound call progress (alerting)
- Send and Receive DTMF digits
- Receive far-end disconnect supervision

Answer Call

- Answer an incoming PBX call
- Receive call party identification (CPID)
- Send and Receive DTMF digits
- Release an active call

Transfer Call

- Transfer a connected PBX call
- Answer an incoming (transferred) PBX call
- Receive call party identification (CPID) during incoming transfer
- Receive DTMF digits after transfer completed

Message Waiting

- Send message waiting notification request (Set)
- Send no message waiting notification request (Clear)

9.3.1.2 Restrictions

The PBX self verification diagnostic is currently only available on PBX interfaces and protocols that allow the interfaces and channels to be individually assigned extension numbers and allows outbound and inbound call routing from one interface and channel to another.

Note: The PBX self verification diagnostics is not available on any ISDN protocol.

9.3.1.3 Capabilities

The ability to route calls and send message waiting status updates through a particular PBX interface and channel is determined by the capabilities, enabled state, and if the interface and channel is in-service. If the interface and channel is not in-service or is not enabled, or it does not support the capability for the requested action, then the interface and channel will not be used for routing through the gateway and will not be included in a diagnostic test.

To view and modify the capabilities and enabled state on a PBX interface and channel navigate to the *Gateway -> Gateway Capabilities* Web page on the Media Gateway.

9.3.1.4 Location

The PBX self verification diagnostic tool is only part of the suite of diagnostic utilities that can be used to assist in recording, testing, and resolving configuration or compatibility issues.

The Media Gateway diagnostic utilities are accessible via the Web interface by selecting the *Diagnostics* link on the left side menu of any Web page.

9.3.2 PBX Self Verification Test Operation

To operate the PBX self verification diagnostic tool, navigate to the *PBX Self Verification* tab via the Media Gateway's user interface.

In the PBX Self Verification Test Configuration table:

1. Choose the diagnostic test to perform by selecting one of the options in the **Test Selection** field. The default selection is the *Initiate Call / Answer Call* diagnostic test only, but all diagnostic test calls can be selected to be executed as one operation.
2. If the *Initiate Call / Answer Call* or *Transfer Call* diagnostic tests are selected, fill in the interface extension numbers of the interfaces that are to be included in the test.
   - Initiate Call / Answer Call - Requires two valid interface extension numbers.
   - Transfer Call - Requires three valid interface extension numbers and is executed with the Initiate Call / Answer Call diagnostic test.
3. If the Send Message Waiting Status diagnostic test is selected, fill in the **Message Waiting Extension Number** field with the extension to send the status update requests.

Note: The *Send Message Waiting Status* diagnostic test does not allow individual interfaces to be selected for inclusion/exclusion. The test will be executed on all available PBX interfaces.

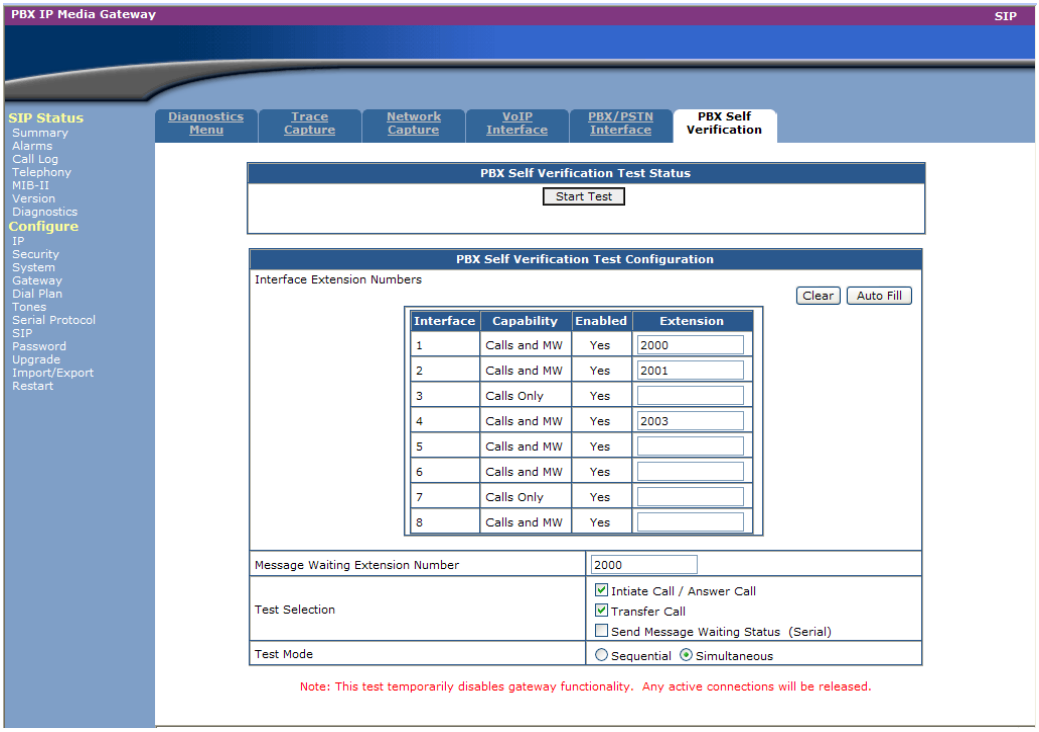
4. Select the **Test Mode** to be *Sequential* or *Simultaneous* (this is an optional parameter for the test).

Note: The *Send Message Waiting Status* diagnostic tests are always executed in Sequential mode due to possible load issues of the PBX under test.

To start the specified diagnostic test, press the **Start Test** button on the Web page.

Note: Executing the PBX/PSTN interface diagnostic tool temporarily disables gateway functionality. Any active connections will be released.

Figure 48. PBX Self Verification Web Page



9.3.2.1 Configuration Options

The following configuration items are available to be modified by the user:

Interface Extension Numbers

Enter the extension number of the interfaces to include in the selected diagnostic test. Interface extension numbers are only valid for *Initiate Call / Answer Call*, and *Transfer Call* diagnostic tests. Interface extension numbers that are left blank will not be included in the diagnostic test.

Message Waiting Extension Number

Enter the extension number to send the message waiting status update requests. Message waiting extension number is only valid for the *Send Message Waiting Status* diagnostic test selection.

Test Selection

Choose the type of diagnostic test to perform. Available options are *Initiate Call / Answer Call, Transfer Call* and *Send Message Waiting Status*.

Test Mode

Select the execution mode of the diagnostic test to be either *Simultaneous* or *Sequential*. Sequential test mode executes only a single diagnostic test at one time. The next diagnostic test will not start until the previous test has completed. Simultaneous test mode executes as many diagnostic tests at once as possible. The next diagnostic test will start as soon as the previous test has started.

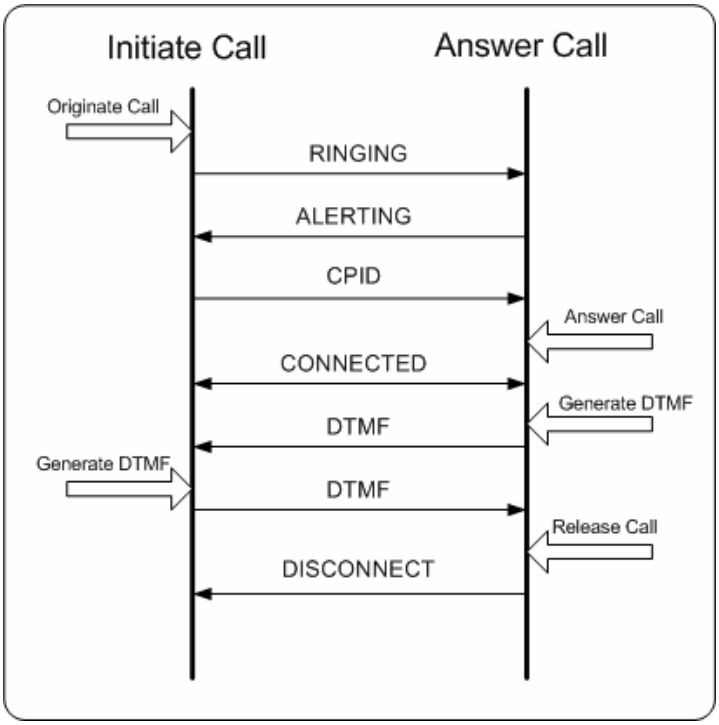
The *Send Message Waiting Status* diagnostic tests are always executed in Sequential mode due to possible load issues of the PBX under test.

9.3.2.2 Call Flows

Initiate Call / Answer Call

The call flow of the *Initiate Call / Answer Call* diagnostic test tries to verify the operation of originating an outbound call and answering an inbound call. These tests also include verification of outbound call progress, inbound CPID, send/receive DTMF and disconnect supervision.

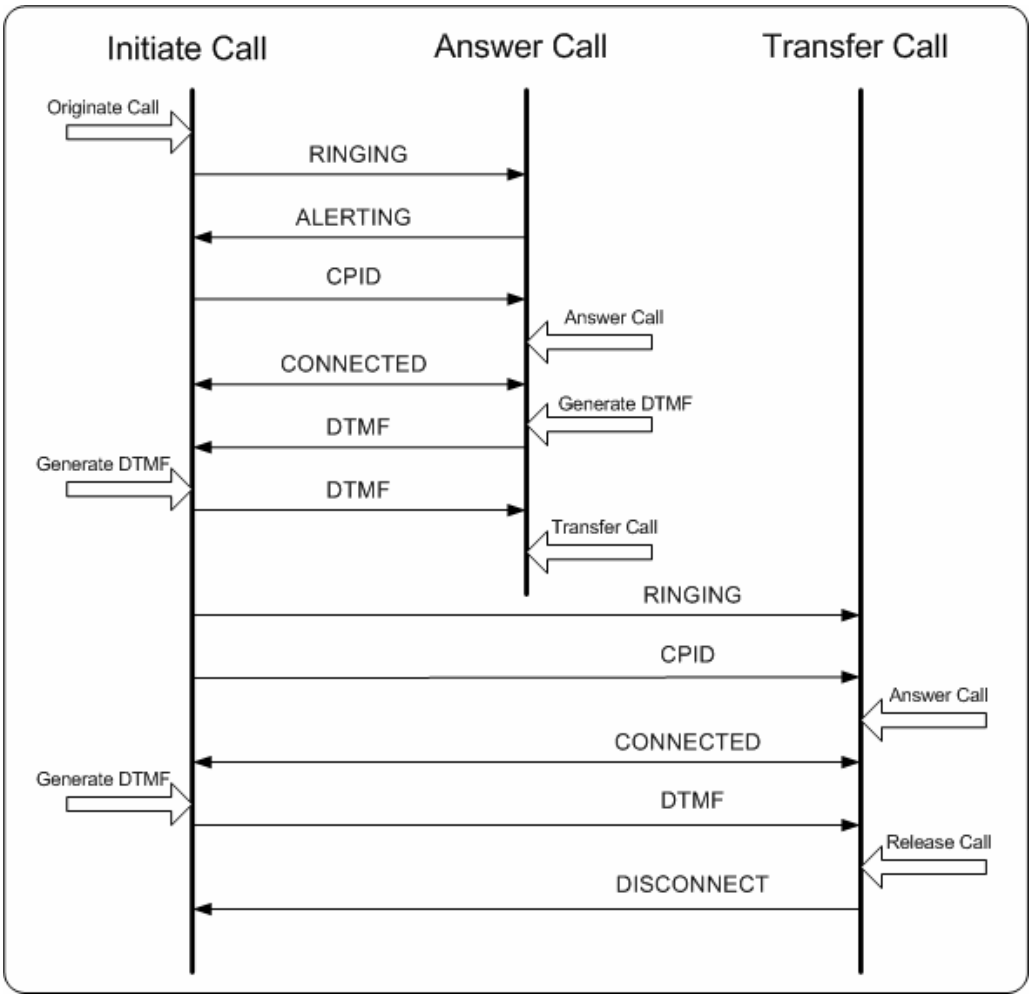
Figure 49. Call Flow for Initiate Call / Answer Call



Initiate Call / Answer Call and Transfer Call

The call flow of the *Initiate Call* / *Answer Call* and *Transfer Call* diagnostic test tries to verify not only the operation of originating an outbound call and answering an inbound call, but transferring the call as well. These tests also include verification of outbound call progress, inbound CPID, send/receive DTMF and disconnect supervision.

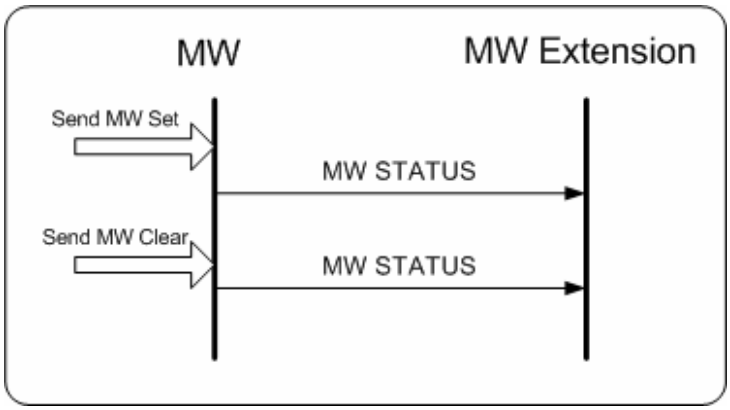
Figure 50. Call Flow for Initiate Call / Answer Call and Transfer Call



Send Message Waiting Status

The call flow of the *Send Message Waiting Status* diagnostic test tries to verify the message waiting notification set and clear update operation.

Figure 51. Call Flow for Send Message Waiting Status



9.3.2.3 Test Status

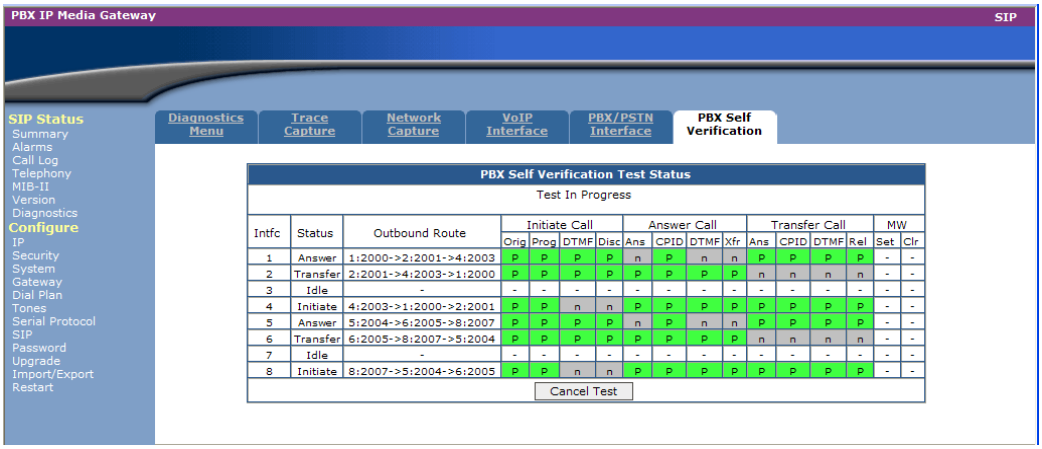
When a diagnostic test is executed the test results are shown in the PBX Self Verification Test Status table on the Web page.

Test Result Symbol	Description
-	Test skipped because not configured or not available (no capability, not enabled, or not in-service).
n	Test was pending but did not execute because a failure occurred.
P	Test Passed
F	Test Failed

The test can be canceled at any time by pressing the **Cancel Test** button.

When the test is completed a hyperlink will be displayed to allow the user to return to the PBX Self Verification Test configuration Web page.

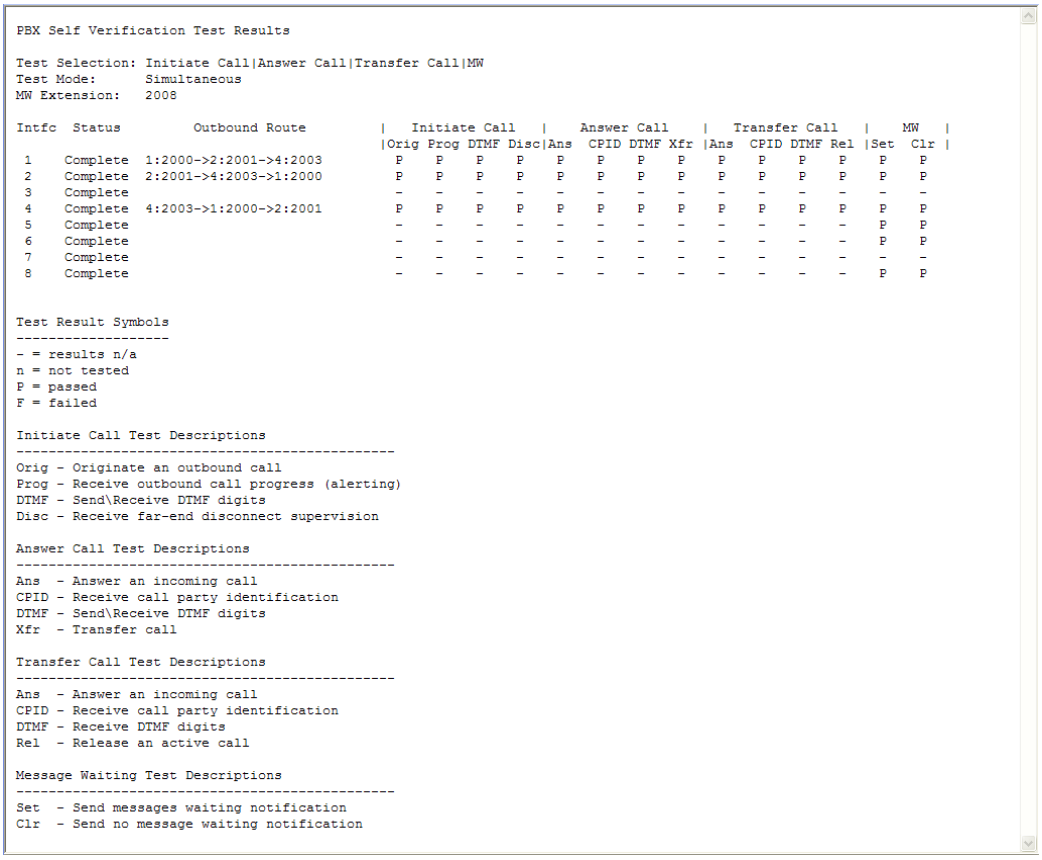
Figure 52. PBX Self Verification Test Status Web Page



9.3.2.4 Test Results

After the diagnostic test(s) are completed or canceled the test results can be opened or saved to a text file (.txt) for documented proof of the verification process. To open the test results in another Web browser window click the ***pbxtest.txt*** hyperlink.

Figure 53. PBX Self Verification Test Results



```
PBX Self Verification Test Results

Test Selection: Initiate Call|Answer Call|Transfer Call|MW
Test Mode:      Simultaneous
MW Extension:   2008

Intfc  Status         Outbound Route         |   Initiate Call   |    Answer Call    |   Transfer Call   |    MW   |
                                             |Orig Prog DTMF Disc|Ans  CPID DTMF Xfr |Ans  CPID DTMF Rel |Set  Clr |
 1     Complete  1:2000->2:2001->4:2003        P    P    P    P    P    P    P    P    P    P    P    P    P    P
 2     Complete  2:2001->4:2003->1:2000        P    P    P    P    P    P    P    P    P    P    P    P    P    P
 3     Complete                                -    -    -    -    -    -    -    -    -    -    -    -    -    -
 4     Complete  4:2003->1:2000->2:2001        P    P    P    P    P    P    P    P    P    P    P    P    P    P
 5     Complete                                -    -    -    -    -    -    -    -    -    -    -    -    P    P
 6     Complete                                -    -    -    -    -    -    -    -    -    -    -    -    P    P
 7     Complete                                -    -    -    -    -    -    -    -    -    -    -    -    -    -
 8     Complete                                -    -    -    -    -    -    -    -    -    -    -    -    P    P


Test Result Symbols
-------------------
- = results n/a
n = not tested
P = passed
F = failed

Initiate Call Test Descriptions
-----------------------------------------------
Orig - Originate an outbound call
Prog - Receive outbound call progress (alerting)
DTMF - Send\Receive DTMF digits
Disc - Receive far-end disconnect supervision

Answer Call Test Descriptions
-----------------------------------------------
Ans  - Answer an incoming call
CPID - Receive call party identification
DTMF - Send\Receive DTMF digits
Xfr  - Transfer call

Transfer Call Test Descriptions
-----------------------------------------------
Ans  - Answer an incoming call
CPID - Receive call party identification
DTMF - Receive DTMF digits
Rel  - Release an active call

Message Waiting Test Descriptions
-----------------------------------------------
Set  - Send messages waiting notification
Clr  - Send no message waiting notification
```

9.3.2.5 Test Failures

If a failure occurs some suggestions are:

- Verify PBX interface is in-service and configured properly
- Verify gateway is configured for specific PBX interface (is there a default .ini file)
- Run the test again with a minimum number of extensions configured and in *Sequential* test mode.
- Run the test again with the **Trace Capture** diagnostic utility started to log the detailed call flow of the PBX self verification diagnostic test.
- Use the **PBX/PSTN Interface** diagnostic utility to test *Initiate Call* and *Send Message Waiting Status* on a particular interface and channel.
- Check the Call Log for the call end reason of the connections used in the diagnostic tests.

Some of the most common test result failures and causes are:

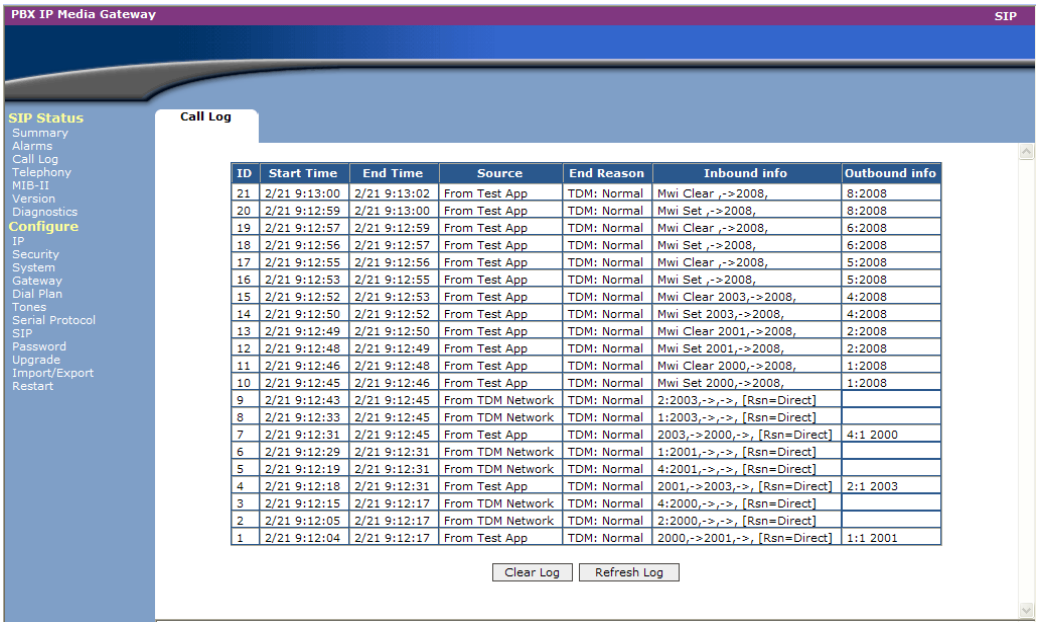
Test	Failure Reasons
Originate an outbound PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*. • Dial tone is not detected or being provided.
Answer an outbound PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*.
Receive outbound call progress (alerting)	• Call progress is not detected or being provided.
Receive call party identification (CPID)	• The switch is not providing CPID. • The programmed CPID type on the Media Gateway does not match the CPID type being provided by the switch (ex. Type I or Type II DMTF)
Send and Receive DTMF digits	• The PBX interface and channel are not configured properly*. • The switch does not allow DTMF between interfaces and channels.
Receive far-end disconnect supervision	• The PBX interface and channel are not configured properly*. • Disconnect tone is not detected or being provided.
Transfer a connected PBX call	• The extension numbers for the PBX interface are not correct. • The PBX interface and channel are not configured properly*. • Dial tone is not detected or being provided.
Send messages waiting notification request	• The PBX interface and channel are not configured properly*. • No confirmation tone from the switch. • Feature Access Codes (FAC) is not correct. • The MWI extension number is not correct or does not accept message waiting status updates. • The MWI uses serial port and/or it is not configured properly.
*For more details on configuration, see the *Installation and Configuration Integration Notes*.	

9.3.2.6 Call Log

The test results from the PBX self verification diagnostic tool are also stored in the Media Gateway call log. The *Source* field will contain **From Test App** to indicate that the connection was created by a diagnostic application.

The call log is accessible through the Web by selecting the *Call Log* link on the left side menu of any Web page.

Figure 54. PBX Self Verification Call Log Web Page



ID	Start Time	End Time	Source	End Reason	Inbound info	Outbound info
21	2/21 9:13:00	2/21 9:13:02	From Test App	TDM: Normal	Mwi Clear ,->2008,	8:2008
20	2/21 9:12:59	2/21 9:13:00	From Test App	TDM: Normal	Mwi Set ,->2008,	8:2008
19	2/21 9:12:57	2/21 9:12:59	From Test App	TDM: Normal	Mwi Clear ,->2008,	6:2008
18	2/21 9:12:56	2/21 9:12:57	From Test App	TDM: Normal	Mwi Set ,->2008,	6:2008
17	2/21 9:12:55	2/21 9:12:56	From Test App	TDM: Normal	Mwi Clear ,->2008,	5:2008
16	2/21 9:12:53	2/21 9:12:55	From Test App	TDM: Normal	Mwi Set ,->2008,	5:2008
15	2/21 9:12:52	2/21 9:12:53	From Test App	TDM: Normal	Mwi Clear 2003,->2008,	4:2008
14	2/21 9:12:50	2/21 9:12:52	From Test App	TDM: Normal	Mwi Set 2003,->2008,	4:2008
13	2/21 9:12:49	2/21 9:12:50	From Test App	TDM: Normal	Mwi Clear 2001,->2008,	2:2008
12	2/21 9:12:48	2/21 9:12:49	From Test App	TDM: Normal	Mwi Set 2001,->2008,	2:2008
11	2/21 9:12:46	2/21 9:12:48	From Test App	TDM: Normal	Mwi Clear 2000,->2008,	1:2008
10	2/21 9:12:45	2/21 9:12:46	From Test App	TDM: Normal	Mwi Set 2000,->2008,	1:2008
9	2/21 9:12:43	2/21 9:12:45	From TDM Network	TDM: Normal	2:2003,->,->, [Rsn=Direct]	
8	2/21 9:12:33	2/21 9:12:45	From TDM Network	TDM: Normal	1:2003,->,->, [Rsn=Direct]	
7	2/21 9:12:31	2/21 9:12:45	From Test App	TDM: Normal	2003,->2000,->, [Rsn=Direct]	4:1 2000
6	2/21 9:12:29	2/21 9:12:31	From TDM Network	TDM: Normal	1:2001,->,->, [Rsn=Direct]	
5	2/21 9:12:19	2/21 9:12:31	From TDM Network	TDM: Normal	4:2001,->,->, [Rsn=Direct]	
4	2/21 9:12:18	2/21 9:12:31	From Test App	TDM: Normal	2001,->2003,->, [Rsn=Direct]	2:1 2003
3	2/21 9:12:15	2/21 9:12:17	From TDM Network	TDM: Normal	4:2000,->,->, [Rsn=Direct]	
2	2/21 9:12:05	2/21 9:12:17	From TDM Network	TDM: Normal	2:2000,->,->, [Rsn=Direct]	
1	2/21 9:12:04	2/21 9:12:17	From Test App	TDM: Normal	2000,->2001,->, [Rsn=Direct]	1:1 2001

9.4 Diagnostic Logging

The following is an overview describing the diagnostic logging capabilities of the Media Gateway and includes the following sections:

- Overview
- Debug Trace Capture
- Network Capture
- TDM Capture

9.4.1 Overview

The Media Gateway provides the user with the ability to record, in real-time, diagnostic information. This information can then be transferred to a PC for off-line analysis of problems.

Three independent logs are available:

- **Debug Trace Capture:** The debug trace log is a verbatim copy of the trace messages that are available using the terminal interface. The messages are described in Section 9.6, “Trace Mechanism”, on page 223.
- **Network Capture:** The network log is a copy of the IP messages transmitted and received by the Gateway.

- **TDM Capture:** The TDM log is a copy of the voice data transferred on the T1/E1 line. It is available only on the DMG2000.

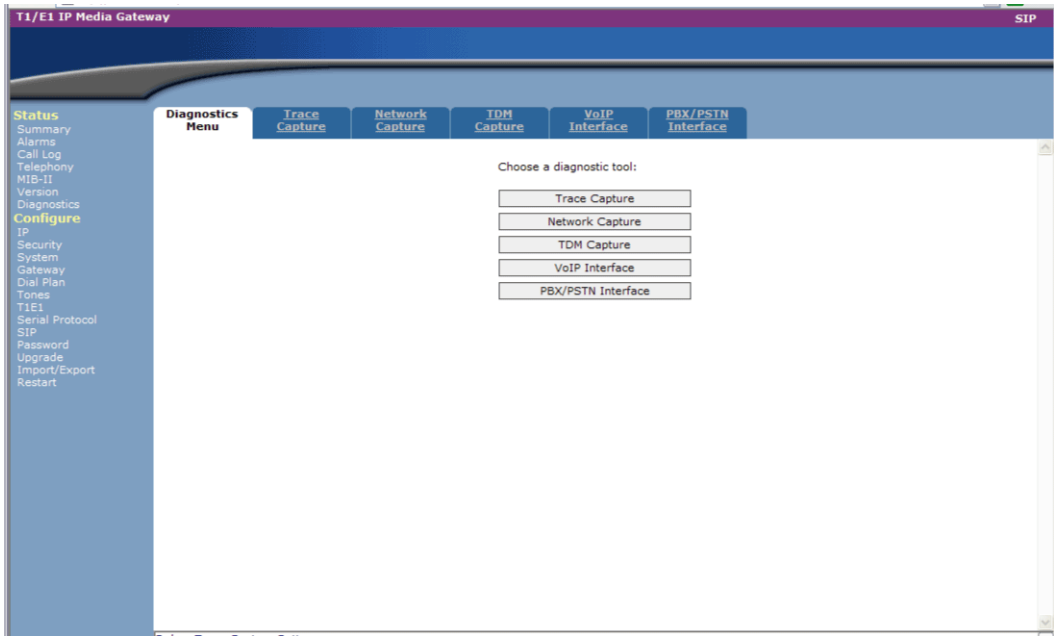
To view a log you must transfer the log to a PC. When transferred to a PC, each log is formatted into an industry standard to facilitate viewing and sharing of data. The format for each log is:

- **Debug Trace Capture**: Uses an ASCII text format (.TXT)
- **Network Capture**: Uses an Ethereal format. Ethereal is a freeware program downloadable from the Internet. It is required only to view Network Captures, not to transfer them from the Media Gateway to a PC.
- **TDM Capture**: Uses a Wave file format (.wav). Wave format is a standard Windows® format and can be opened by programs such as Windows Media Player, Cool Edit and Adobe Audition. None of these programs are required to transfer the file from the Gateway to a PC, but one is required to view or playback the file.

9.4.1.1 Control of Log Files:

The control of each log file is independent (i.e. more than one log can be simultaneously created) and is administered via the Media Gateway's Web interface. You reach the logging pages by using the Diagnostic Page. The diagnostic Web page is shown in Figure 55.

Figure 55. Diagnostic Web Page



In summarizing the overview of Diagnostic Logging, remember:

- Logging is available on Media Gateway Version 5.0 Software and later.
- The DMG1000 provides two logs: Debug Trace capture and Network capture.
- The DMG2000 provides three logs: Debug Trace capture, Network capture and TDM capture.

- The Media Gateway stores the logs in RAM. Therefore, the logs are not preserved if the unit is reset or powered down. To save a log you must transfer it to a PC.
- Special applications are not required to download logs from the Media Gateway to a PC, but are required to view the logs.
- All control of logging is through the Web interface.

9.4.2 Debug Trace Capture

The Debug Trace Capture enhances the trace mechanism described in Section 9.6, "Trace Mechanism", on page 223. Specifically, it allows users to set the trace mask via the Web page in addition to allowing the trace output to be redirected to a log file that can then be downloaded. The DMG2000 Debug Trace Capture control page is shown in Figure 56. While the DMG2000 control page has more trace keys, the ideas presented in this section apply equally to both the DMG1000 and the DMG2000.

Figure 56. Debug Trace Capture Control Page - DMG2000

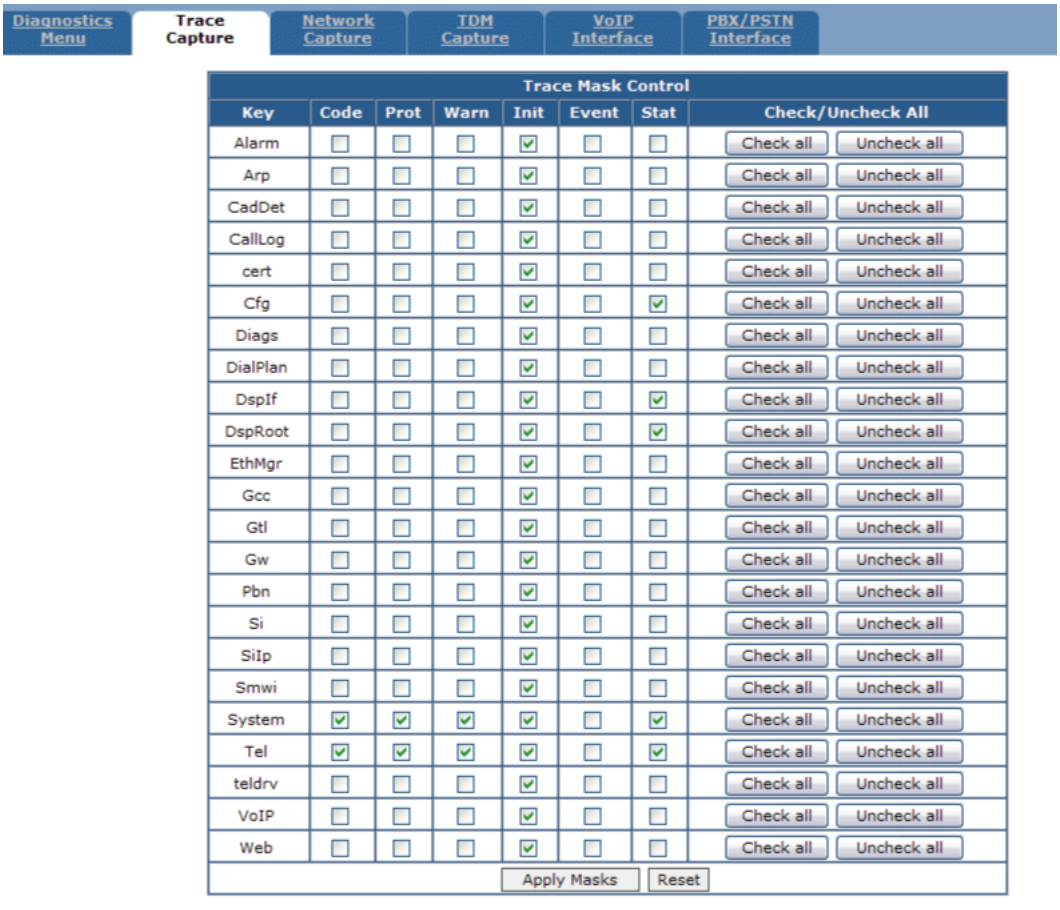
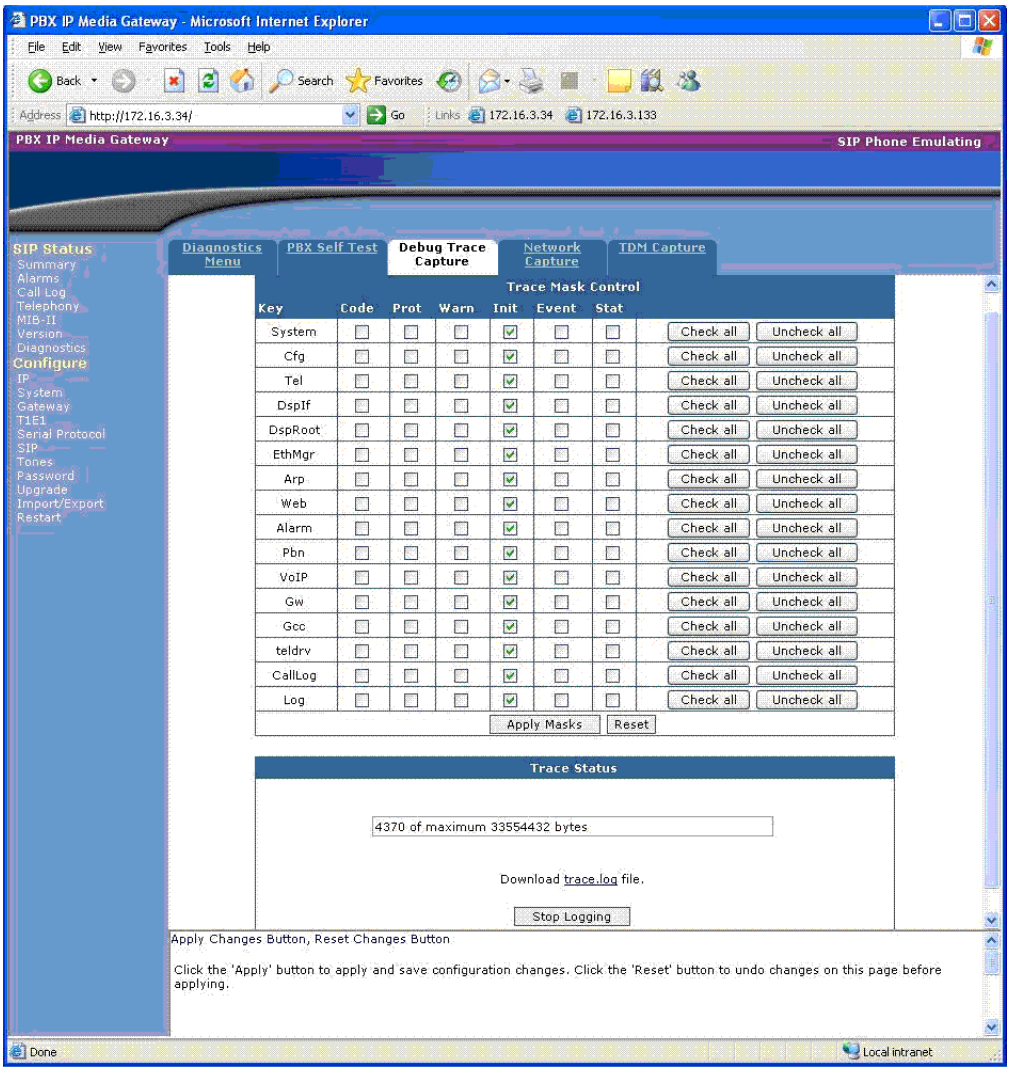


Figure 56 shows the default trace keys (i.e. immediately after power-up). Notice the Error and Debug trace masks are not shown as they are always enabled.

To use the Debug Trace Capture, first check the appropriate trace masks. Click **Apply Masks** and then click **Start Logging**. Logging starts and the display changes to show the state of the log. An example of a running log is shown in Figure 57.

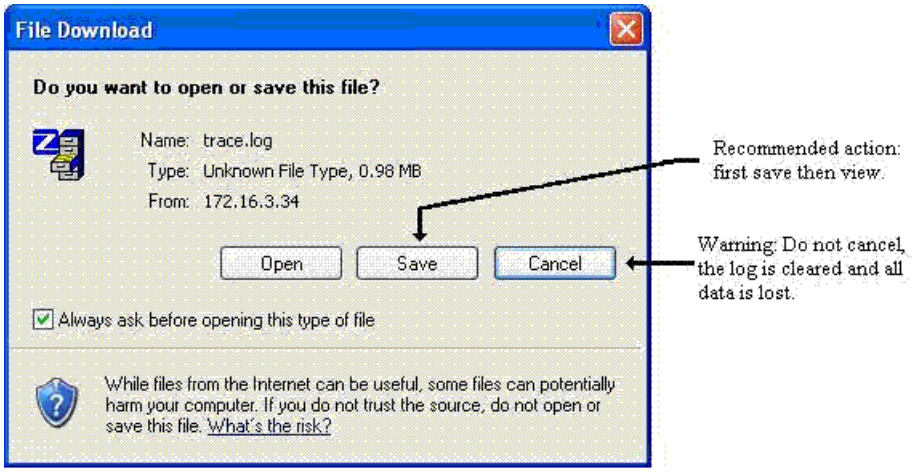
Figure 57. Example of a Running Log - Debug Trace Capture



When the Debug Trace Capture is running, the fill level of the log is displayed. The log uses a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity, the fill level does not change.

The log can be transferred to a PC by right-clicking **Download trace.log file**. Doing this brings up a File Download dialog box as shown in Figure 58. Select **Save** to save the file to your PC. After saving, you can view the file using most any ASCII text editor, such as Wordpad or Notepad.

Figure 58. File Download Dialog Box for Trace.log


File Download
Do you want to open or save this file?
Name: trace.log
Type: Unknown File Type, 0.98 MB
From: 172.16.3.34
Open
Save
Cancel
Always ask before opening this type of file
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?
Recommended action: first save then view.
Warning: Do not cancel, the log is cleared and all data is lost.

Important points to remember about Debug Trace Capture are:

- After modifying the trace keys be sure to click Apply Masks.
- It is not necessary to stop logging to change trace keys. The keys can be changed on the fly.
- Downloading the Debug Trace Capture clears the log and starts a new log file.
- Stopping the log and then restarting it does not clear the log, new data is just appended to the log.
- The DMG1000 log can store up to 1.5 Mbytes, while the DMG2000 log can store up to 32 Mbytes.
- Debug Trace Capture is automatically enabled at power-on-with Error, Debug and Init trace masks for all trace keys.

9.4.3 Network Capture

The Network Capture creates an in-memory log of Media Gateway IP traffic. Since there can be a great deal of traffic, the Network Capture control page allows filtering of IP packets. Filtering packets makes it easier to analyze problems by reducing the number of irrelevant messages. It also increases the time before the log wraps, thus making it more probable an error will be caught.

9.4.3.1 Network Capture Details

The DMG1000 unit has one network interface:

- LAN1: an external network that connects the DMG1000 to a VOIP network

The DMG2000 unit has three network interfaces:

- DSP: an internal network that connects the DMG2000 DSP to the DMG2000 CPU
- LAN1: an external network that connects the DMG2000 to a network (VOIP)
- LAN2: an external network that connects the DMG2000 to a network (non-VOIP)

Notes: ***1.*** The DMG1000's LAN1 is equivalent to the DMG2000's LAN1.

2. Currently, LAN2 is only supported in Version 5.1 SU1 Software or later.

The Network Capture is controlled from the Media Gateway's Web interface. The DMG2000 Network Capture control page is shown in Figure 59. Even though the DMG1000 control page has only one network interface (LAN1), the ideas presented in this section apply equally to the DMG2000 and the DMG1000.

Figure 59. Network Capture Control Page - DMG2000

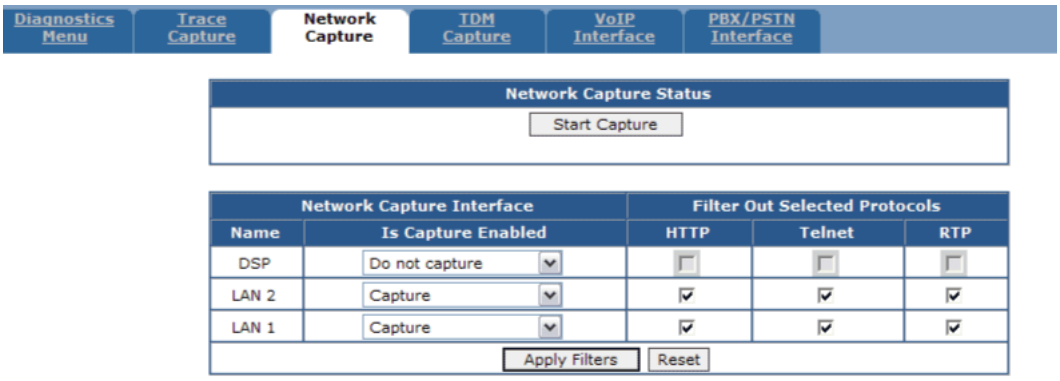
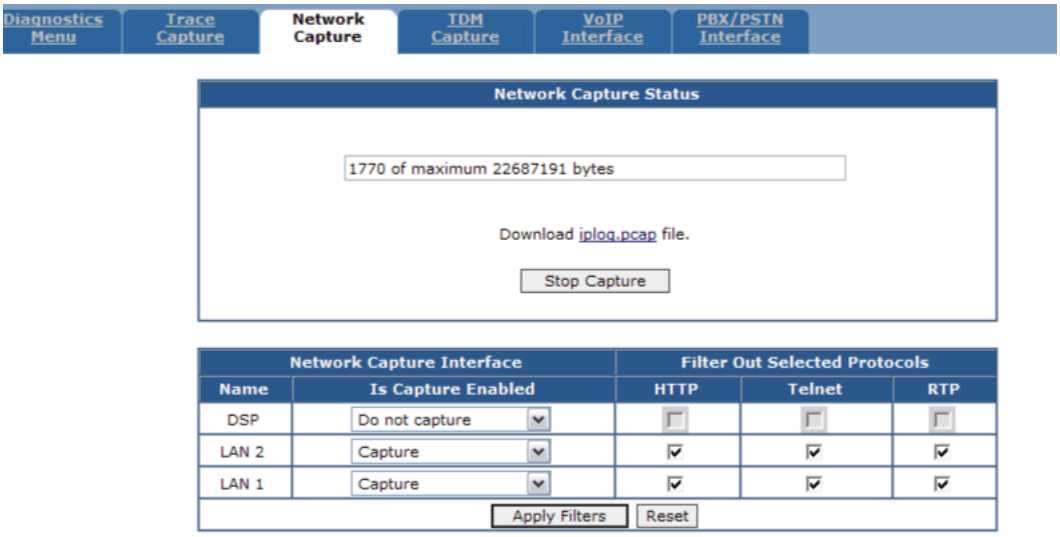


Figure 59 shows the default values (i.e. immediately after power-up) for a DMG2000. The first level of filtering occurs on the interface level. You can select **Capture** to enable or **Do not capture** to disable individual interfaces. The second level of filtering is on the protocol level. When an interface is enabled (Capture), you have the ability to filter (not log) different protocols. A check mark in the box means the Gateway will not log packets of the corresponding protocol.

To use the Network Capture, first select the interfaces to capture, check the protocols you do not want to log, click **Apply Filters** and then click **Start Capturing**. Logging starts and the display changes to show the state of the log. An example of a running log is shown in Figure 60.

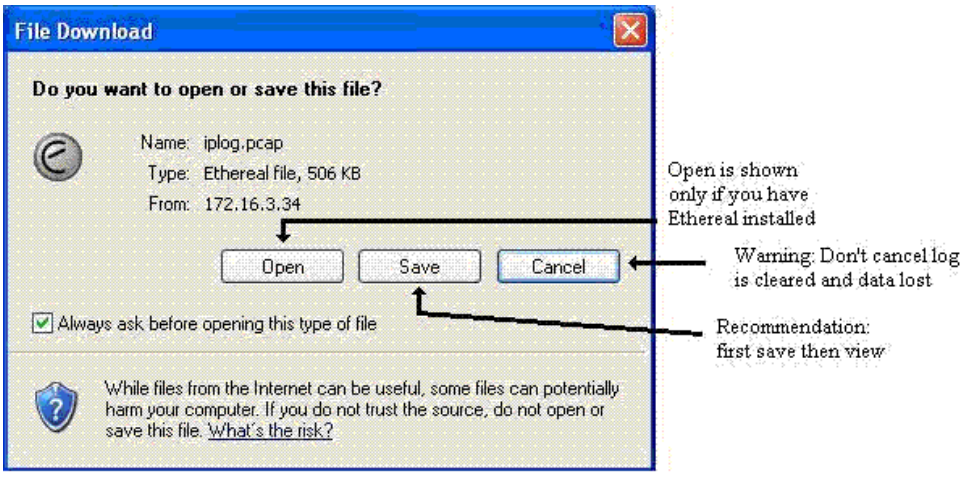
Figure 60. Example of Running Log - Network Capture



When the Network Capture is running, the fill level of the log is displayed. The log uses a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity the fill level does not change.

The log can be transferred to a PC by right-clicking **Download iplog.pcap file**. Doing this brings up a File Download dialog box as shown in Figure 61. Select **Save** to save the file to the PC. After saving, you can view the file with any network capture program that supports libpcap format (e.g. Ethereal).

Figure 61. File Download Dialog Box for Iplog.pcap


File Download
Do you want to open or save this file?
Name: iplog.pcap
Type: Ethereal file, 506 KB
From: 172.16.3.34
Open
Save
Cancel
Always ask before opening this type of file
While files from the Internet can be useful, some files can potentially harm your computer. If you do not trust the source, do not open or save this file. What's the risk?
Open is shown only if you have Ethereal installed
Warning: Don't cancel log is cleared and data lost
Recommendation: first save then view

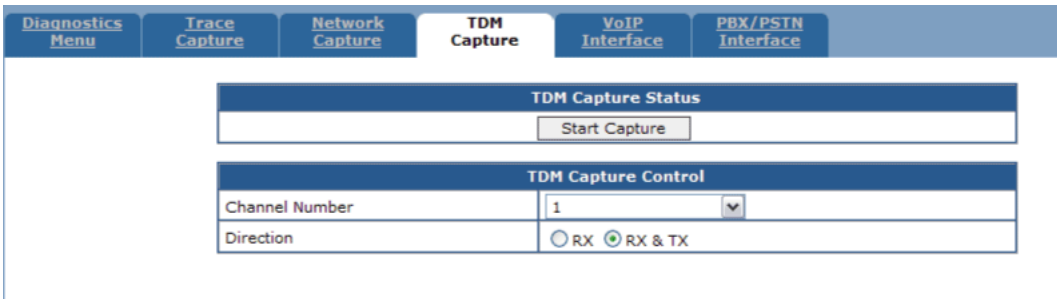
Important points to remember about Network Capture are:

- Downloading does not require any special software, but viewing requires Ethereal.
- On the Network Capture Web page, checking a protocol type causes it NOT to be captured.
- After selecting protocol filters, be sure to click Apply Filters.
- Size of the log is determined at startup. DMG1000 ~1.1 Mbytes, DMG2000 ~24 Mbytes.
- Downloading the Network Capture clears the log and starts another.
- Stopping the log and restarting it does not clear the log, new data is just appended.

9.4.4 TDM Capture

The TDM Capture creates an in-memory log of voice traffic on the DMG2000's T1 or E1 line. The capture is controlled from the DMG2000's Web interface, as shown in Figure 62. You can log voice traffic from a single channel in either the Rx direction, or the Rx and Tx direction, where Rx is defined as entering the DMG2000 and Tx as leaving the DMG2000. The TDM Capture is not supported by the DMG1000.

Figure 62. TDM Capture Control Web Page



9.4.4.1 TDM Capture Details

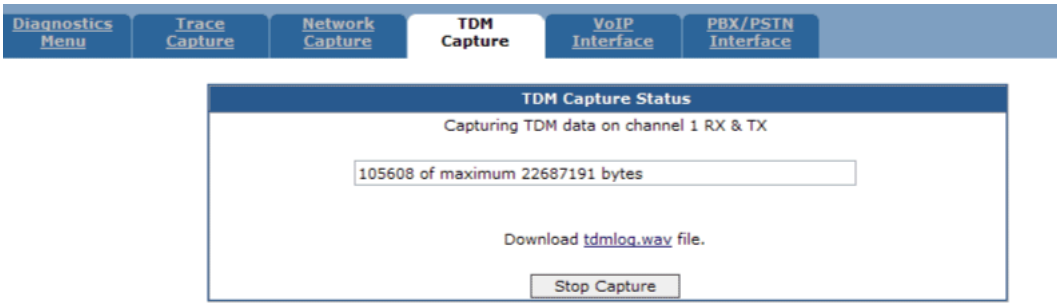
Figure 62 shows the default values (i.e. immediately after power-up) for the TDM Capture. You can select any one channel for logging. The range of channels is dependent on the TDM configuration (T1 or E1) and the number of hardware spans. Table 10 shows the mapping from a protocol and span to the TDM Capture channel numbers.

Table 10. Mapping of Protocol and Span Numbers to TDM Capture Channel Numbers

	Span 1	Span 2	Span 3	Span 4
T1 CAS	1 to 24	25 to 48	49 to 72	73 to 96
T1 QSIG	1 to 23	24 to 46	47 to 69	70 to 92
T1 NI-2	1 to 23	24 to 46	47 to 69	70 to 92
E1 QSIG	1 to 30	31 to 60	61 to 90	91 to 120

To use the TDM Capture, enter the Channel Number to capture, select either RX or RX & TX, then click **Start Capturing**. Logging begins and the display changes to show the state of the log. An example of a running log is shown in Figure 63.

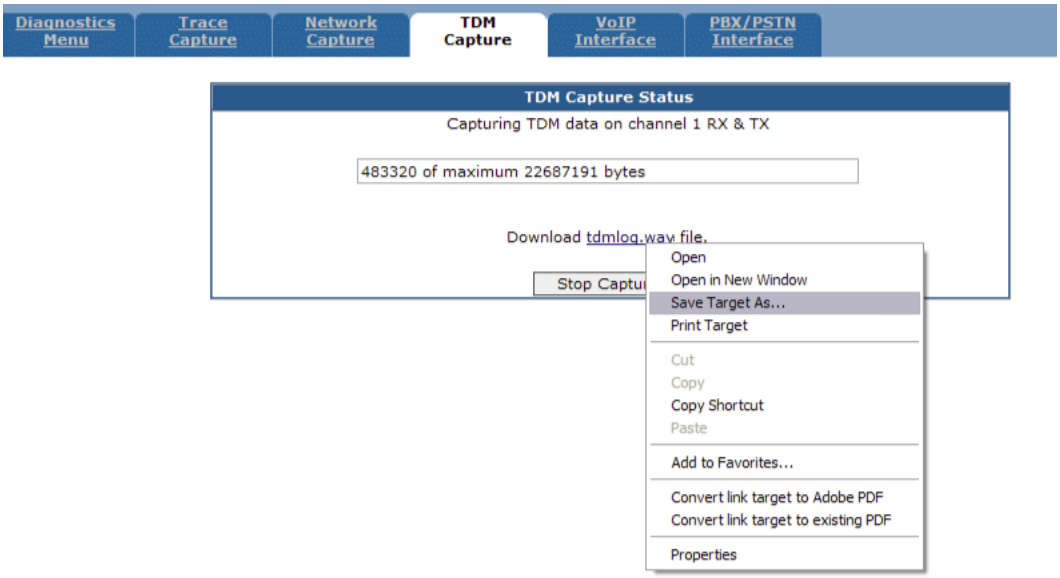
Figure 63. Example of Running Log - TDM Capture



While the TDM Capture is running, the fill level of the log is displayed. The log is a circular buffer, so when it reaches capacity, new data overwrites the old. Once the log reaches capacity, the fill level does not change.

The log can be transferred to a PC by right clicking on **Download tdmlog.wav file** and selecting **Save Target As…** from the drop-down menu, Figure 64. After saving, you can play the file using Windows Media Player or view it with a variety of audio file viewers (ex. CoolEdit Pro, Adobe Audition).

Figure 64. File Download Dialog for Tdmlog.wav



Important points to remember about TDM Capture are:

- For proper downloading you must right click on Download tdmlog.wav file.

  Note: A right click is necessary due to the Windows® OS mapping the .wav extension to Windows Media Player. Said mapping results in a left click automatically bringing up Windows Media Player, which does not allow for saving a file.

- TDM capture occurs regardless of channel call state.
- Capture times
  - Rx only = ~ 48 minutes (data into DMG2000)
  - Rx & Tx = ~ 24 minutes
- Downloading does not require any special software, but viewing requires an audio viewer such as CoolEdit Pro or Adobe Audition (not freeware).
- Downloading the TDM Capture clears the log and starts another.
- Stopping the log and restarting clears the log. **Caution:** This operation is different from the other logs.

9.5 Communicating to the Terminal Interface

Provided you have Admin level privileges, you may communicate with the Media Gateway terminal interface by connecting to the serial interface (DIAGNOSTICS connector on DMG1000 types and COM2 connector on DMG2000 types) located on the rear panel or by establishing a telnet session via the LAN port.

Information in this section includes:

- Connecting to Terminal Interface Via DIAGNOSTICS Connector
- Connecting to Terminal Interface Via LAN Connector

9.5.1 Connecting to Terminal Interface Via DIAGNOSTICS Connector

Perform the following:

1. Connect a serial cable to the serial interface connector on the rear panel of the Media Gateway. For connector pin designation information, refer to the DIAGNOSTICS Connector Pin Designations table (DMG1000) or COM1 and COM2 Connector Pin Designations table (DMG2000) in the *Getting Started Guide*.
2. Using a standard serial interface application (for example, Procomm Plus or HyperTerminal), set the workstation to the following:
   - Baud Rate = 38400 for DMG1000, 115200 for DMG2000
   - Parity = None
   - Data Bits = 8
   - Stop Bits = 1
   - Hardware Flow Control = Off

3. Press the Enter key repeatedly until the following prompt appears:

   PIMG>

4. At the prompt, type **pwd** and press Enter.
5. When prompted, enter the password for the admin user (the default is **IpodAdmin**) and press Enter.

   The Media Gateway will respond with **Admin level accepted**, and then display the prompt. You may now issue terminal commands to the Media Gateway.

9.5.2 Connecting to Terminal Interface Via LAN Connector

Perform the following to initiate a Telnet session at the workstation connected to the Media Gateway via the LAN interface:

1. From the command prompt, type **telnet** followed by a space and the IP address of the Media Gateway. Then press Enter.
2. The **telnet client connected** message will appear. Press Enter. The following prompt will appear:

   PIMG>

3. At the prompt, type **pwd** and then press Enter.
4. When prompted, enter the password for the Admin user (the default is **IpodAdmin**) and press Enter.

   The Media Gateway will respond with **Admin level accepted**, and then display the prompt. You may now issue terminal commands to the Media Gateway.

9.6 Trace Mechanism

The trace mechanism has been designed to provide the maximum flexibility in tracing the various modules of the Media Gateway software. The trace utility is supported by the serial port terminal as well as the telnet terminal, and is available to users at Admin levels and above. Therefore, to enable the trace utility, the user must log on to the terminal using a valid password.

Note: The trace mechanism can be accessed via the Debug Trace Capture interface as described in Section 9.4.2, “Debug Trace Capture”, on page 215.

Information in this section includes:

- Trace Format
- Trace Utility
- Trace Commands
- Examples of Trace Commands and Displays

The trace function allows the user to enable or disable different types of traces in different software modules. It also allows the user to enable or disable traces that are related to the telephony ports on the Media Gateway. Therefore, control of telephony port traces is separate from the control of traces of the various software modules. This allows the user to set the trace control of the modules as desired, and then separately set the telephony ports that are needed to be traced.

Traces that are related to a telephony port number are tagged with the 1-based telephony port number to which they are related. For instance:

Telephony port 1 events:

```
002:53:542 [Tel-1    ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:53:782 [Tel-1    ] Event    0:0:48|            250         |
002:53:784 [Tel-1    ] Event    Cpid (250->) (Direct)
```

Telephony port 2 events:

```
002:59:314 [Tel-2     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:59:550 [Tel-2     ] Event    0:0:48|            251         |
002:59:552 [Tel-2     ] Event    Cpid (251->) (Direct)
```

Time-stamping of the trace output can be enabled or disabled by the user.

The telephony switch protocol can be traced, but the output is encoded for security.

9.6.1 Trace Format

The following is a description of the format of a trace display. The example display is broken down into its components.

Example of trace display:

```
003:02:372 [Tel-2     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
```

Components:

003:02:372 - time-stamp in minutes:seconds:milliseconds

[Tel-2] - trace-key and telephony port (no port if non port-related)

Event - trace type

Lamp 60:CallApp0:0 FLASH->OFF

9.6.2 Trace Utility

The trace utility separates the control of software module trace output using trace keys. A single trace key represents a functional software module. The trace utility then uses trace types that can be enabled or disabled for each trace key. The trace keys and trace types are used to enable or disable traces using the trace utility.

9.6.2.1 Trace Key

The trace key defines the specific sub-system within the Media Gateway to trace. For example, if you are only interested in capturing SIP packets (SIP Emulating or SIP Driving mode), you would use the VoIp trace key:

```
trace voip <trace type> on
```

Table 11 describes all trace keys supported by the trace function as well as the trace types available to each trace key.

Table 11. Supported Trace Keys

Trace Keys	Description	Prot	Code	Error	Warn	Init	Alarm	Event	Stat
System	OS-Independent layer, System initialization		●	●	●	●	●		
Tel	Digital Telephony Interface (for protocol, hook, lamp, display, CPID, in phone-emulation and phone-driving etc.)	●(8)	●	●	●	●	●	●(6)	
TelDrv	Low layer ISDN trace coming from the Telesoft driver. Includes the RAW ISDN layer 3 packets. (14)	●(15)	●	●	●	●	●	●	
VoIp	Voice Over IP Software Stacks	●(1)	●(10)			●			
DspCpi	Dsp Media/Call Progress API (media session setup, RTP stats)		●	●	●	●	●	●	●(7)
DspIf	DSP Media Interface (14)								
CadDet	Dsp Call Progess (14)		●	●	●	●		●	
DspRoot	DSP OS-independent layer		●	●	●				
DspVad	DSP Voice Activity Detector		●	●	●			●	
DspDmi	DSP Management Interface		●	●	●				
DspDlmm	DSP Management Interface		●	●	●				
DspNw	Network Interface for DSP (13)		●	●	●				
DspUdp	UDP/IP Protocol Stack handling for DSP (13)		●	●	●				
DspIarp	OS independent ARP interface (13)		●	●	●				
Arp	ARP Interface for DSP (14)			●					
EthMgr	Ethernet Manager for DSP (14)		●	●		●			
Gcc	Generic Call Control API		●	●	●	●			
Gw	Gateway Application State Machine		●	●	●	●	●		
Cfg	Configuration Module		●	●	●	●	●		
Web	Web Interface		●	●	●	●	●		
Si	Serial Interface API (2)	●(9)	●	●	●	●			
Smwi	Serial Interface MWI Device (2)		●	●	●	●			
SiIp	Serial Interface IP Layer (2)		●	●	●	●	●		
Pbn	Packet-Network Interface		●	●	●	●	●		
H323if	H.323 State Machine		●	●	●	●	●		
SipCsm	SIP State Machine		●	●	●	●	●		
Adept	Adept display parsing		●	●	●	●			
CallLog	Call Log Handler							●	
Alarm	Alarm Handler		●	●	●	●			

Table 11. Supported Trace Keys (Continued)

Trace Keys	Description	Prot	Code	Error	Warn	Init	Alarm	Event	Stat
Web	Web Server		●	●	●				
Log	Diagnostics Logging Module (aka Enhanced Diagnostics)		●	●	●				
Phd	Phone-Driving GCC Layer (3)		●	●	●	●	●		
PhnDrv	Phone-Driving State Machine (3)		●	●	●	●	●		
iNimDrv	NIM (telephony network) Driver (3,4)	●(11)	●	●	●	●	●		
Dcif	Direct-Connect Interface (5)		●	●	●	●	●		
NimCtlDc	NIM/DirectConnect Interface (5)		●	●	●	●	●		
PBXTest	PBX Self Test Application (12)		●	●	●	●		●	
LTA	LearnTone Application (12)		●	●	●	●		●	

NOTES:
1) Currently SIP only - enable to trace SIP packets
2) Serial Interface API only active if serial mode enabled
3) Active only if in phone-driving mode
4) Active only on AMI DMG1000
5) Active only in Direct-Connect mode
6) enable to see telephony hookswitch, lamps, voice state, CPID, displays, tone detection (call progress and dtmf)
7) enable to trace RTP packet and error statistics during a media session
8) enable to trace telephony protocol packets (encoded in release version)
9) enable to trace serial telephony interface packets (serial server only)
10) enable to trace H.323 and SIP Trillium code
11) enable to trace iNim/DMG1000 interface protocol
12) Active only if feature is run
13) DMG1000 only
14) DMG2000 only
15) RAW ISDN layer 3 messages

Note: H.323 is only supported in Version 5.1 SU1 Software or earlier.

9.6.2.2 Trace Type

Trace type defines the level of tracing. For example, to capture only errors for the VoIp trace key, the trace type would be set to:

```
trace voip error on
```

The following table describes the trace types that are supported:

Table 12. Supported Trace Types

Trace Types	Trace Description
Prot	Protocol
Code	Software/Code
Error	Errors (1)
Warn	Warnings
Init	Software/Device initialization (1)
Alarm	Alarms (1)
Event	Hookswitch, Lamps, Tones, Displays, Button Presses
Stat	Statistics
(1) System starts with these enabled on all trace keys.	

9.6.3 Trace Commands

The trace commands include the following:

- trace show
- trace default
- trace <trace key> <trace type|all> <on|off>
- trace all <trace type|all> <on|off>
- trace <trace key|all> reset
- trace all off
- trace port <1-based port number|all> <on|off>
- trace time <on|off>
- trace -l

9.6.3.1 trace show

The trace show command displays the trace levels of all trace modules and port-related trace settings. The following is an example of a trace show display:

```
PIMG>trace show
Key          Traces
----------   ------------------------
System       Error Init Debug
Cfg          Error Init Debug
Tel          Error Init Debug
DspIf        Error Init Debug
DspRoot      Error Init Debug
EthMgr       Error Init Debug
Arp          Error Init Debug
Web          Error Init Debug
Alarm        Error Init Debug
```

```
Pbn          Error Init Debug
VoIP         Error Init Debug
Gw           Error Init Debug
Gcc          Error Init Debug
Adept        Error Init Debug
teldrv       Error Init Debug
CallLog      Error Init Debug
Log          Error Init Debug

Channel Trace State ('*' enabled, '-' disabled)
-----------------------------------------------
* * * * * * * *
```

9.6.3.2 trace default

The trace default command sets the default tracing of all trace keys, and enables VoIP and telephony event tracing.

```
PIMG>trace default
```

9.6.3.3 trace <trace key> <trace type|all> <on|off>

This trace command enables or disables trace types on specified trace keys.

Use:

```
trace tel all on
```

to enable all telephony traces

Use:

```
trace tel event on
```

to enable telephony control traces

9.6.3.4 trace all <trace type|all> <on|off>

The trace all command enables/disables trace types on all trace keys.

Use:

```
trace all code off
```

to disable all code traces on all trace keys

Use:

```
trace voip code on
```

to enable code traces only on voip stacks

9.6.3.5 trace <trace key|all> reset

This command resets specified (or all) trace keys to their defaults (from system start).

9.6.3.6 trace all off

This command disables all traces, except **Error** and **Debug**, on all trace keys. **Error** and **Debug** cannot be disabled.

9.6.3.7 trace port <1-based port number|all> <on|off>

This command enables or disables port-related traces.

Use:

```
trace port all off
```

to disable all telephony-port related traces

Use:

```
trace port 1 on
```

to enable only telephony port 1 related traces

9.6.3.8 trace time <on|off>

The trace time command enables or disables time stamps on traces.

9.6.3.9 trace -l

This command displays trace-types.

9.6.4 Examples of Trace Commands and Displays

The following are some examples of various trace commands and the resulting displays:

Example 1

Enable tracing of telephony events, such as button presses, lamps, displays, tones, etc.:

```
PIMG>trace tel event on
Ok
PIMG>exit
Good-Bye.
002:53:542 [Tel-1     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:53:782 [Tel-1     ] Event    0:0:48 |                        250          |
002:53:784 [Tel-1     ] Event    Cpid (250->) (Direct)
```

```
002:59:314 [Tel-2     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
002:59:550 [Tel-2     ] Event    0:0:48 |                        251          |
002:59:552 [Tel-2     ] Event    Cpid (251->) (Direct)
003:02:300 [Tel-1     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
003:02:372 [Tel-2     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
003:02:490 [Tel-1     ] Event    0:0:48 |          MAY 13  7:44 P             |
003:02:570 [Tel-2     ] Event    0:0:48 |          MAY 13  7:44 P             |
```

Example 2

Enable tracing of SIP messages:

```
PIMG>trace voip prot on
Ok
PIMG>exit
Good-Bye.
[Tel-1     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
[Tel-1     ] Event    0:0:48 |                        250                    |
[Tel-1     ] Event    Cpid (250->) (Direct)
[VoIP      ] Prot     <----v=0!
[VoIP      ] Prot          o=phone 26533 0 IN IP4 10.0.1.110!
[VoIP      ] Prot          s=-!
[VoIP      ] Prot          c=IN IP4 10.0.1.110!
[VoIP      ] Prot          t=0 0!
[VoIP      ] Prot          m=audio 49010 RTP/AVP 0 8 101!
[VoIP      ] Prot          a=rtpmap:0 PCMU/8000/1!
[VoIP      ] Prot          a=rtpmap:8 PCMA/8000/1!
[VoIP      ] Prot          a=rtpmap:101 telephone-event/8000!
[VoIP      ] Prot          a=fmtp:101 0-15!
[VoIP      ] Prot          a=sendrecv!
[VoIP      ] Prot
[VoIP      ] Prot     <----INVITE sip:101@10.0.1.215;Transport=udp SIP/2.0!
[VoIP      ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP      ] Prot          To:sip:101@10.0.1.215!
[VoIP      ] Prot          Content-Type:application/SDP!
[VoIP      ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP      ] Prot          CSeq:1 INVITE!
[VoIP      ] Prot          Allow-Events:refer,message-summary!
[VoIP      ] Prot          Expires:120!
[VoIP      ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP      ] Prot          Contact:<sip:port1@10.0.1.110:5060>!
[VoIP      ] Prot          User-Agent:DMG1000 !
[VoIP      ] Prot          Max-Forwards:70!
[VoIP      ] Prot          Supported:100rel,timer!
[VoIP      ] Prot          Content-Length:216!
[VoIP      ] Prot          !
[VoIP      ] Prot          v=0!
[VoIP      ] Prot          o=phone 26533 0 IN IP4 10.0.1.110!
[VoIP      ] Prot          s=-!
[VoIP      ] Prot          c=IN IP4 10.0.1.110!
[VoIP      ] Prot          t=0 0!
[VoIP      ] Prot          m=audio 49010 RTP/AVP 0 8 101!
[VoIP      ] Prot          a=rtpmap:0 PCMU/8000/1!
[VoIP      ] Prot          a=rtpmap:8 PCMA/8000/1!
[VoIP      ] Prot          a=rtpmap:101 telephone-event/8000!
[VoIP      ] Prot          a=fmtp:101 0-15!
[VoIP      ] Prot          a=sendrecv!
```

```
[VoIP      ] Prot
[VoIP      ] Prot     ---->SIP/2.0 100 Trying!
[VoIP      ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP      ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP      ] Prot          CSeq:1 INVITE!
[VoIP      ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP      ] Prot          To:sip:101@10.0.1.215;tag=3246313536412A29002455E5!
[VoIP      ] Prot          Contact:<sip:101@10.0.1.215:5060>!
[VoIP      ] Prot          Server:DMG1000 1.2!
[VoIP      ] Prot          Supported:100rel,timer!
[VoIP      ] Prot          Content-Length:0!
[VoIP      ] Prot          !
[VoIP      ] Prot
[VoIP      ] Prot     ---->SIP/2.0 180 Ringing!
[VoIP      ] Prot          Via:SIP/2.0/UDP 10.0.1.110:5060!
[VoIP      ] Prot          Call-ID:01B2270D9C81400000000006@pbxgw.default.com!
[VoIP      ] Prot          CSeq:1 INVITE!
[VoIP      ] Prot
From:"250"<sip:port1@10.0.1.110:5060>;user=phone;tag=1A1F32463135364100000ABE!
[VoIP      ] Prot          To:sip:101@10.0.1.215;tag=3246313536412A29002455E5!
[VoIP      ] Prot          Contact:<sip:101@10.0.1.215:5060>!
[VoIP      ] Prot          Server:DMG1000 1.2!
[VoIP      ] Prot          Supported:100rel,timer!
[VoIP      ] Prot          Content-Length:0!
[VoIP      ] Prot          !
[VoIP      ] Prot
```

Example 3

Enable trace of RTP statistics:

```
PIMG>trace dspcpi stat on
Ok
PIMG>exit
Good-Bye.
[DspCpi-1  ] Stat     Receive/Transmit Stats:
[DspCpi-1  ] Stat         Rx Voice Packets:             0
[DspCpi-1  ] Stat         Tx Voice Packets:             2
[DspCpi-1  ] Stat         Tx Silence Suppressed Frames: 0
[DspCpi-1  ] Stat         Rx Min Jitter:                -1 (ms)
[DspCpi-1  ] Stat         Rx Max Jitter                 0 (ms)
[DspCpi-1  ] Stat         Rx RTP Avg Jitter:            0 (pcm samples)
[DspCpi-1  ] Stat         Tx Grant Sync Dropped Frames: 0
[DspCpi-1  ] Stat         Tx Octets:                    160
[DspCpi-1  ] Stat         Rx Octets:                    0
[DspCpi-1  ] Stat         AAL2 Coding Profile Changes:  0
[DspCpi-1  ] Stat         DTMF Tx Packets:              0
[DspCpi-1  ] Stat         DTMF Rx Packets:              0
[DspCpi-1  ] Stat         SID Rx Packets:               0
[DspCpi-1  ] Stat         SID Tx Packets:               0
[DspCpi-1  ] Stat         Tx Last Timestamp:            10313
[DspCpi-1  ] Stat         Tx Extended Seq Number:       0
[DspCpi-1  ] Stat         Tx Last Seq Number(AAL2-UUI): 1
[DspCpi-1  ] Stat         Tx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Timestamp:            0
[DspCpi-1  ] Stat         Rx Last Ssrc:                 0
```

```
[DspCpi-1  ] Stat         Rx Extended Seq Number:       0
[DspCpi-1  ] Stat         Rx Last Seq Number(AAL2-UUI): 0
[DspCpi-1  ] Stat         Pkt Lost by Network:          0
[DspCpi-1  ] Stat         Rx P2P Packets to Hosts:      0
[DspCpi-1  ] Stat         Rx P2P Filtered Packets:      0
[DspCpi-1  ] Stat         P2P Squelched Voice Packets:  0
[DspCpi-1  ] Stat         Rx Net Packets:               0
[DspCpi-1  ] Stat         Tx Net Packets:               2
[DspCpi-1  ] Stat     Error Stats:
[DspCpi-1  ] Stat         invalid_header_count:         0
[DspCpi-1  ] Stat         to_micro_overflow_count:      0
[DspCpi-1  ] Stat         lost_enh_packet_count:        0
[DspCpi-1  ] Stat         no_core_packet_count:         0
[DspCpi-1  ] Stat         pkt_lost_by_network:          0
[DspCpi-1  ] Stat         rc4key_update_lost_pkt_count: 0
[DspCpi-1  ] Stat         invalid_mac_header_count:     0
[DspCpi-1  ] Stat         invalid ssrc count:           0
[DspCpi-1  ] Stat         invalid payload count:        0
[DspCpi-1  ] Stat     Voice Playout Stats:
[DspCpi-1  ] Stat         avg_playout_delay:            20
[DspCpi-1  ] Stat         lost_packet_count:            0
[DspCpi-1  ] Stat         replay_packet_count:          0
[DspCpi-1  ] Stat         idle_packet_count:            0
[DspCpi-1  ] Stat         dropped_packet_count:         0
[DspCpi-1  ] Stat         rx_packet_count:              0
[DspCpi-1  ] Stat         avg_frame_jitter:             2 (ms)
[DspCpi-1  ] Stat         adpt_po_buf_delay_inc:        0
[DspCpi-1  ] Stat         adpt_po_buf_delay_dec:        0
[DspCpi-1  ] Stat         po_buf_underflow_cnt:         0
[DspCpi-1  ] Stat         cell_starve_evt_cnt:          0
[DspCpi-1  ] Stat     Receive/Transmit Stats:
[DspCpi-1  ] Stat         Rx Voice Packets:             151
[DspCpi-1  ] Stat         Tx Voice Packets:             506
[DspCpi-1  ] Stat         Tx Silence Suppressed Frames: 0
[DspCpi-1  ] Stat         Rx Min Jitter:                30 (ms)
[DspCpi-1  ] Stat         Rx Max Jitter                 30 (ms)
[DspCpi-1  ] Stat         Rx RTP Avg Jitter:            0 (pcm samples)
[DspCpi-1  ] Stat         Tx Grant Sync Dropped Frames: 0
[DspCpi-1  ] Stat         Tx Octets:                    40480
[DspCpi-1  ] Stat         Rx Octets:                    36480
[DspCpi-1  ] Stat         AAL2 Coding Profile Changes:  0
[DspCpi-1  ] Stat         DTMF Tx Packets:              0
[DspCpi-1  ] Stat         DTMF Rx Packets:              0
[DspCpi-1  ] Stat         SID Rx Packets:               0
[DspCpi-1  ] Stat         SID Tx Packets:               0
[DspCpi-1  ] Stat         Tx Last Timestamp:            50633
[DspCpi-1  ] Stat         Tx Extended Seq Number:       0
[DspCpi-1  ] Stat         Tx Last Seq Number(AAL2-UUI): 505
[DspCpi-1  ] Stat         Tx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Pkt Type:             0
[DspCpi-1  ] Stat         Rx Last Timestamp:            67547
[DspCpi-1  ] Stat         Rx Last Ssrc:                 1368282060
[DspCpi-1  ] Stat         Rx Extended Seq Number:       0
[DspCpi-1  ] Stat         Rx Last Seq Number(AAL2-UUI): 151
[DspCpi-1  ] Stat         Pkt Lost by Network:          0
[DspCpi-1  ] Stat         Rx P2P Packets to Hosts:      0
[DspCpi-1  ] Stat         Rx P2P Filtered Packets:      0
[DspCpi-1  ] Stat         P2P Squelched Voice Packets:  0
[DspCpi-1  ] Stat         Rx Net Packets:               152
[DspCpi-1  ] Stat         Tx Net Packets:               506
```

```
[DspCpi-1  ] Stat     Error Stats:
[DspCpi-1  ] Stat         invalid_header_count:         1
[DspCpi-1  ] Stat         to_micro_overflow_count:      0
[DspCpi-1  ] Stat         lost_enh_packet_count:        0
[DspCpi-1  ] Stat         no_core_packet_count:         0
[DspCpi-1  ] Stat         pkt_lost_by_network:          0
[DspCpi-1  ] Stat         rc4key_update_lost_pkt_count: 0
[DspCpi-1  ] Stat         invalid_mac_header_count:     0
[DspCpi-1  ] Stat         invalid ssrc count:           1
[DspCpi-1  ] Stat         invalid payload count:        0
[DspCpi-1  ] Stat     Voice Playout Stats:
[DspCpi-1  ] Stat         avg_playout_delay:            27
[DspCpi-1  ] Stat         lost_packet_count:            0
[DspCpi-1  ] Stat         replay_packet_count:          0
[DspCpi-1  ] Stat         idle_packet_count:            0
[DspCpi-1  ] Stat         dropped_packet_count:         0
[DspCpi-1  ] Stat         rx_packet_count:              453
[DspCpi-1  ] Stat         avg_frame_jitter:             1 (ms)
[DspCpi-1  ] Stat         adpt_po_buf_delay_inc:        0
[DspCpi-1  ] Stat         adpt_po_buf_delay_dec:        0
[DspCpi-1  ] Stat         po_buf_underflow_cnt:         0
```

Example 4

Enable trace of telephony protocol:

```
PIMG>trace tel prot on
Ok
PIMG>exit
Good-Bye.
[Tel-1     ] Prot
PIMGXX00152BB986F7A64E5028A7A1E69B2C7FA0D56681CB1D9F92D287C80C3F5D96593
[Tel-1     ] Event    Lamp 60:CallApp0:0 OFF->FLASH
[Tel-1     ] Prot
PIMGXX01171850BBCFB4660E32E3C0FC67C53DF45CB602FC847DCF07E52F678A1E8153EA9
[Tel-1     ] Prot     PIMGXX010744A
[Tel-1     ] Prot
PIMGXX021790EDBC5B8E6F74C62762B8BE5CD5A11DDE49E8BD7AACA72910DAE88013BB77D
[Tel-1     ] Prot     PIMGXX020610FA12D05
[Tel-1     ] Prot
PIMGXX0310E4CD136BB199A9C4596613EFB98084BB5EF810030EBEE7A1A8CD5CBA1A81831
[Tel-1     ] Prot     PIMGXX0307F2B
[Tel-1     ] Prot
PIMGXX041E8AA914C95D9F88E5210FA3395984BD28D205B67AC46B4281E0AE991BE6606
[Tel-1     ] Event    0:0:48 |                        250                    |
[Tel-1     ] Event    Cpid (250->) (Direct)
[Tel-1     ] Prot
PIMGXX0511126EB93134DD19188F9BE6A62682B82F6D4083E8DBBB1C5D7ABAF688009BC
[Tel-1     ] Prot
PIMGXX0612911CB3EE87834C804848D1611B468DAA9496CEF645A59429ADC64505C5BE8FB
[Tel-1     ] Prot     PIMGXX0605CCB
[Tel-1     ] Prot
PIMGXX071EF122ABB171D6D0CD98074220E19E8901C84B68AE601B1372923FE2255E19E
[Tel-1     ] Event    Lamp 60:CallApp0:0 FLASH->OFF
[Tel-1     ] Prot
PIMGXX081DF265B1B82C6B322456F4E979ABF50492CDE81A1CC65EE64FDD407E29D08A5
[Tel-1     ] Event    0:0:48 |          MAY 13  7:46 P                      |
```

9.7 Diagnostic Commands

This section describes a number of diagnostic commands and includes:

- Devstat Command
- Restart Command
- Ping Command
- Ver Command
- Alarm List Command

9.7.1 Devstat Command

The Devstat command displays the current high-level settings of the unit and the unit's main status and telephony port status. The telephony port state can be 'In Service', 'Out of Service', or 'Active'. 'Active' means that the port is off-hook. 'In Service' means that the port is available for use, but is currently not in use. 'Out of Service' means that the port is disconnected.

```
PIMG>devstat
Description       Value
=====================================================================
sysStartType      0x0
MAC               00-a0-e6-02-01-19
IP                10.0.1.110
Subnet Mask       255.255.255.0
Gateway           10.0.1.1
Host              10.0.1.63
Status            OK

Port 1            Active
Port 2            In Service
Port 3            In Service
Port 4            In Service
Port 5            In Service
Port 6            In Service
Port 7            In Service
Port 8            In Service
```

9.7.2 Restart Command

The Restart command causes the unit to warm-boot.

```
restart
```

9.7.3 Ping Command

The Ping command generates ICMP requests to the specified IP address. This is useful in determining connection status to IP endpoints, gateways, or routers.

```
PIMG>ping 10.0.1.1
10.0.1.1 is alive
PIMG>ping 10.0.1.1 5
PING 10.0.1.1: 56 data bytes
64 bytes from 10.0.1.1: icmp_seq=0. time=2. ms
64 bytes from 10.0.1.1: icmp_seq=1. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=2. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=3. time=0. ms
64 bytes from 10.0.1.1: icmp_seq=4. time=0. ms
----10.0.1.1 PING Statistics----
5 packets transmitted, 5 packets received, 0% packet loss
round-trip (ms) min/avg/max = 0/0/2
PIMG>ping 10.10.10.10
no answer from 10.10.10.10
```

9.7.4 Ver Command

The Ver command outputs the versions of all hardware and firmware of the unit.

```
PIMG>ver
Description       Version
=====================================================================
DSP               |9.1| |THU AUG 29 15:22:54 2002|
Flash App         |lab_D| |THU NOV 21 16:20:47 2002|
Flash DSP         |9.1| |THU AUG 29 15:22:54 2002|
Boot              |dhcp| |WED NOV 20 12:48:16 2002|
CPLD              0x21
iNIM App          6.6
iNIM FPGA         179
iNIM Boot         5.0
Base HW           2
```

9.7.5 Alarm List Command

The Alarm List command outputs the current list of generated system alarms. The timestamp of each alarm is the time since unit start in hours:minutes:seconds.

```
PIMG>alarms
000:00:000        System            MIB-II Trap: Cold start (0)
000:00:000        System            MIB-II Trap: Cold start (0)
000:12:000        Telephony         snw (1) - In Service
000:12:000        Telephony         snw (1) - In Service
000:12:000        Telephony         snw (2) - In Service
000:13:000        Telephony         snw (4) - In Service
000:13:000        Telephony         snw (5) - In Service
000:13:000        Telephony         snw (6) - In Service
000:13:000        Telephony         snw (7) - In Service
000:13:000        Telephony         snw (8) - In Service
000:15:000        Telephony         snw (3) - In Service
```

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